



REASON

Version
2.5

→ **Operation Manual**

14 Channel Expandable Mixer → Analog Polysynth → Grainable Polysynth → Digital Samplers → REX-loop Player → Drum Machine → ReBirth Input Device → Multiple Effects Processors → Shelving and Parametric EQs → Master Song Sequencer → Pattern Sequencer → 64 Channel Audio Output → 64 Channel ReWire Output → 512 Band Vocoder → CV Processing Tools → Full Automation → Total Recall

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REASON

1

→ The Sequencer

Recording

Recording and Playback Basics

The basic procedures for recording and playback are described in the *Getting Started* book. Here is a brief repetition:

- **To activate recording, click the Record button on the Transport panel or press [*] on the numeric keypad.**
If your computer keyboard lacks a numeric keypad, you can activate recording by holding down [Command] (Mac) or [Ctrl] (Windows) and pressing [Return].
- **Recording starts at the current song position.**
- **You can get a metronome click during recording by activating Click on the transport panel.**
The volume of the metronome click can be adjusted with the Level knob.
- **If the Loop is activated, the area between the Left and Right locators will be repeated, allowing you to add or replace material on each loop pass (depending on the Overdub/Replace switch - see below).**
- **To start playback from the current song position, click the Play button or press [Enter] on the numeric keypad.**
To stop, click the Stop button, press [0] on the numeric keypad or press [Return]. You can also toggle between Play and Stop by pressing the Space bar.
- **To move the song position, click in the ruler, use the Rewind/Fast Forward buttons or edit the position numerically on the transport panel.**
You can also move the song position to the Left or Right Locator by pressing [1] or [2] on the numeric keypad.

Recording Notes

The Overdub/Replace switch

If you record over an area where there are notes recorded already, the result depends on the Overdub/Replace switch on the transport panel:

- **In Overdub mode, the new recording is added to whatever was on the Track before.**
For example, this is useful for adding elements during loop recording or for adding controller data to recorded notes.
- **In Replace mode, the new recording replaces any previously recorded notes.**
Only the notes in the actual recording area are replaced.
- ! **It's probably best to use Overdub mode as your default mode, to avoid removing material by mistake.**

Quantizing during Recording

If the Quantize Notes During Recording switch is activated on the sequencer toolbar, notes will automatically be quantized when you record them. This is described in detail on [page 29](#).

Recording Controllers

In Reason, you can automate virtually all device parameters, creating completely automated mixes if you like. This is done by recording (or drawing) controllers in the sequencer.

Before you record automation

Before you start recording automation of a parameter, you may want to set it to a suitable "static value". By this, we mean the value the parameter should have whenever it isn't automated in the song. Here is why:

- **When you first record a section of automation for a parameter, its original value will be inserted throughout the rest of the song.**
This is explained in detail on [page 10](#).

Let's say you want to create a fade-out by recording your lowering a fader in the Mixer. Then it's a good idea to first set the fader to the correct static value (i.e. the value the fader should be set to before you start the fade-out).

The same thing is true if you want to create a filter sweep for a synthesizer, somewhere within the song: First set the filter frequency to the value it should have elsewhere in the song, then record the filter sweep.

Recording automation of a device parameter

1. Make sure there is a sequencer track for the device.

For the instrument devices and the Matrix, sequencer tracks are automatically added when you create the device. For a mixer or effect device, you need to add tracks manually, by selecting “Create Sequencer Track for...” from the device context menu. You can also select “Sequencer Track” from the Create menu and connect the created track manually to the desired device (in the Out column in the track list).

2. Click in the In column for the track in the track list, so that a MIDI connector symbol appears.

This indicates that the track will receive MIDI and is ready for recording.

3. Start recording from the desired position.

4. During recording, adjust the desired parameter(s), from the device panel or from a MIDI controller.

You can record automation for several parameters in the same recording pass. However:

→ You can only record automation for one device at a time (the device whose track receives MIDI).

If you want to record automation for parameters on another device, you need to click in the In column for the corresponding track, so that the MIDI connector symbol is moved there.

5. Stop recording.

On the device panel, each automated parameter will have a green frame.



The parameters Feedback and Pan are automated.



In the Arrange view, recorded controllers are indicated in blue (the pale blue “strip” indicates that the track contains any kind of controller automation).

If you play back the recorded section again, the parameters will change automatically. Outside the recorded section, the parameters will have their original settings (the values they had before you started recording).

Recording more for the same controller

If you need to redo a section of recorded automation, or add some automation of the same controller elsewhere in the song, proceed as follows:

! The Overdub/Replace switch does not affect controller recording. However, you should make sure this is set to “Overdub”, to avoid accidentally deleting any notes on the track.

1. Set up and start recording in the same way as described above.

As long as you don’t touch the parameter, its automation data will be played back normally.

2. At the desired position, adjust the parameter.

As soon as you start changing the parameter value, the Punched In indicator will light up on the transport panel.



From this point on, the previously recorded automation will be replaced!

3. Stop recording when you are done.

You have now replaced the section from where you Punched In up to where you stopped recording.

→ Any time after Punching In, you can click the Reset button below the Punched In indicator.

This turns off the Punched In indicator and “resets” the controller recording (making the previously recorded automation active again, from that position). You are still in record mode, and as soon as you adjust the parameter again, the Punched In indicator will be lit.

Basically, clicking the Reset button is the same as stopping recording and starting recording again.

Moving Automated Controllers during Playback - “Live Mode”

Even if you have automated a parameter, you can still “grab it” and adjust it during playback, overriding the automation. This can be very useful when playing Reason live, for example:

1. During playback, click and drag an automated parameter.

The Punched In indicator lights up on the transport panel. From this point on, the recorded automation for the parameter is disabled.

2. To activate the automation again, click the Reset button.

This returns control of the parameter to the sequencer.

→ Automation override is automatically reset when you stop playback.

Background: How recorded controllers are handled

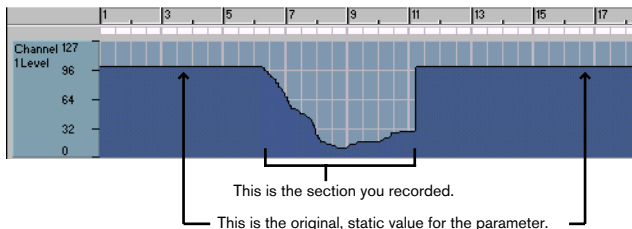
Even though the recording procedures are practically the same, the sequencer handles controllers differently from notes. While each recorded note is a separate event, there are no “controller events” as such in the sequencer. Instead, it works like this:

Each sequencer track has a number of controller “subtracks” (one for each automatable parameter in the corresponding device). A controller subtrack can be viewed as a length of magnetic tape, which you can fill with controller data.

When you haven't yet recorded any automation for a parameter, its subtrack is empty. The parameter is not automated.



As soon as you record anything for the controller, anywhere in the song, the *whole subtrack* is filled with controller data:



This makes it possible to set up a static mix first, and then add some automated parameter changes anywhere in the song while maintaining the static values elsewhere in the song.

Recording Pattern Changes

If your song contains pattern devices, you probably want to use more than a single pattern throughout the song. To facilitate this you can record pattern changes in the sequencer (or draw them in manually, as described on [page 29](#)).

1. **If you want to use the same pattern for the main part of the song (and only want to add some variation patterns here and there), make sure this “main pattern” is selected before you start recording.**

When you first record a pattern change somewhere in the song, the originally selected pattern will be inserted throughout the rest of the song. This is similar to the way controller automation works - see [page 8](#).

2. **Locate the sequencer track for the device, and make sure MIDI is routed to the track.**

That is, the MIDI connector symbol should be shown in the In column for the track in the track list.

3. **Start recording from the desired position.**

When playback starts, the pattern device will automatically start (provided the pattern section is enabled on the device).

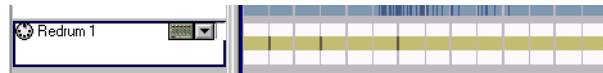
4. **During recording, change patterns with the Bank and Pattern buttons on the device panel.**

Make sure to change the patterns slightly in advance - the actual pattern change will be recorded (and happen) on the next downbeat according to the main sequencer time signature.

5. **When you are done, stop recording.**



The green frame around the pattern buttons indicates that pattern changes are automated for the device.



In the Arrange view, recorded pattern changes are indicated as dark yellow bars (the pale yellow “strip” indicates that the track contains any pattern change data at all).

- **Each pattern change will be recorded on a downbeat (at the start of a new bar in the sequencer).**

You can move pattern changes to other positions in the Edit View if needed (see [page 30](#)).

- **You can punch in on recorded pattern changes, to replace a section of the track.**

This works the same as punching in on controllers (see [page 9](#)).

- **After recording the pattern changes, you can use the function Convert Pattern Track to Notes, to transfer the notes in the patterns to the main sequencer.**

This allows you to create unlimited variations by later editing the notes in the Edit View. See below.

Copying REX loops and Patterns to Sequencer Tracks

As described on [page 188](#), you need to use the “To Track” function when using the Dr.Rex Loop Player device. This creates sequencer notes on the selected track, so that each slice in the REX loop gets a corresponding sequencer note. Playing back the sequencer track will then play all slices in the correct order, with the original timing of the loop.

Similar functions are available for the pattern devices (Redrum and Matrix).

- By using the function Copy Pattern to Track on the Edit menu or device context menu, you can copy the contents of the current pattern to notes on the selected sequencer track.
- The function Convert Pattern Track to Notes works in a similar way, but converts all patterns in a song to notes (taking pattern changes into account).

The procedures differ slightly for the different device types:

Using the “To Track” function for REX Loops

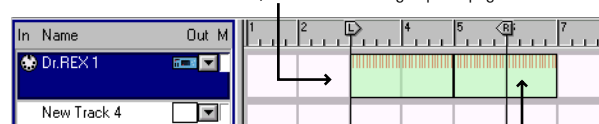
This assumes that you have loaded a REX loop into the Dr.Rex device. For details, see [page 187](#).

1. **Set the left and right locator to encompass the section you want to “fill” with notes for the REX loop.**
2. **Select the track connected to the Dr.Rex device.**
To avoid confusion, make sure there are no events between the locators on the track.
3. **Click the “To Track” button on the Dr.Rex device panel.**
Notes are created for the slices in the loop, and added to the track.



- **If the length of the area between the locators is greater than the length of the REX loop, the loop will be repeated on the track.**
This function always creates an exact number of loops, meaning that the last loop may “stick out” after the right locator.

The created notes are automatically grouped (as indicated by the colored boxes). Read more about groups on [page 18](#).



Here, the loop was 2 bars long. Since there are three bars between the locators, the second loop will stick out after the right locator.

The “Copy Pattern to Track” function

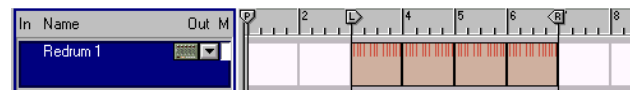
This is available for the Redrum and the Matrix. It is useful when you have created a single pattern and want to use it as starting point for editing in the sequencer. You could also use this if you e.g. have created a drum pattern and want to have this pattern play back some other type of device.

Proceed as follows:

1. **Set the left and right locator to encompass the section you want to “fill” with the notes in the pattern.**
You may want to make sure the length of the area between the locators is a multiple of the pattern length, to avoid “cutting off” the pattern.
2. **Select the track connected to the pattern device.**
In fact, you can select any track. For example, if the device is a Matrix, it may make sense to copy the notes not to the Matrix track, but to the track for the device *controlled* by the Matrix (since the Matrix doesn’t produce any sound in itself, and thus can’t be played by the sequencer notes).
3. **Select the device and select “Copy Pattern to Track” from the Edit menu or the device context menu.**

- **If you selected a track not connected to the pattern device, an alert will appear, asking if that’s really what you want.**
Click OK to proceed, or Cancel to abort.

The pattern is converted to sequencer notes on the track (see the notes below). If the length of the area between the locators is greater than the pattern length, the pattern will be repeated to fill out the area.

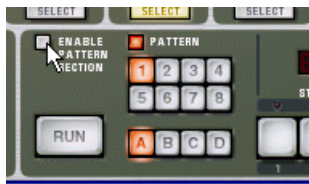


The created notes are automatically grouped (as indicated by the colored boxes). Read more about groups on [page 18](#).

Redrum notes

When you use this function with the Redrum, you should note the following:

- The notes will have the pitch of the corresponding drum sound (see “Using Redrum as a Sound Module” on page 97) and the velocity depending on the Dynamic value.
Soft notes have velocity 30, medium notes have velocity 80 and hard notes have velocity 127.
- You probably want to turn off the “Enable Pattern Section” switch on the Redrum device panel.
Otherwise, the drum sounds will be “double-triggered” when you start playback (once by the pattern section itself, once by the main sequencer).



Matrix notes

When you use this function for the Matrix, you should note the following:

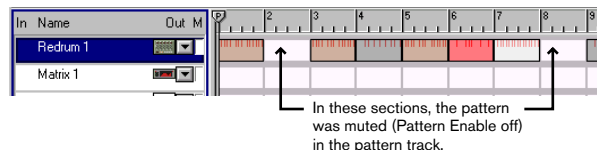
- A note will be created for each pattern step with a gate value other than zero.
The notes will have the pitch according to the key CV value for the step, and the velocity according to the gate value.
- The curve CV is not copied.
- Make sure that the track is connected to the correct device!
Having the track connected to the Matrix itself is pointless, since the Matrix cannot produce any sound.
- You may want to disconnect or even remove the Matrix after performing a “Copy Pattern to Track”.
This is because you probably don't want both the Matrix and the sequencer notes to play back at the same time.

The “Convert Pattern Track to Notes” function

If you have recorded or drawn pattern changes on a Redrum or Matrix track, you can have the whole track converted to notes, in the following way:

1. **Select the track with the pattern changes.**
2. **Select “Convert Pattern Track to Notes” from the Edit menu or the context menu for the track.**

For each bar, the corresponding pattern is converted to notes on the track (following the same rules as for the “Copy Pattern to Track” function). The track will play back just the same as when you played the pattern device with the pattern changes (including the Pattern Enabled/Mute switch).



- **All pattern changes are automatically removed from the track after the operation.**

Redrum notes

- The “Enable Pattern Section” switch is automatically turned off when you use this function.

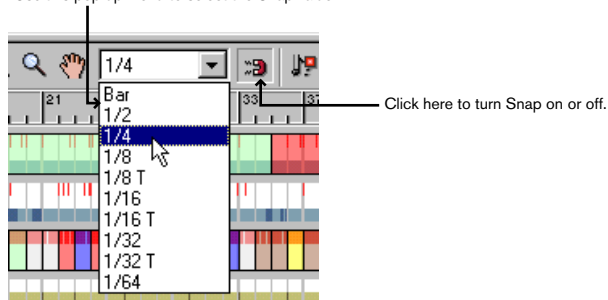
Matrix notes

- After performing “Convert Pattern Track to Notes”, you need to move the contents to another track, or re-route the track to another device.
Having the track connected to the Matrix itself is pointless, since the Matrix cannot produce any sound.
- You may want to disconnect or even remove the Matrix after performing this function.
This is because you probably don't want both the Matrix and the sequencer notes to play back at the same time.

Editing - About Snap

When you select and edit material (both in the Arrange View and the Edit View), the Snap (Snap to Grid) function determines the result. By activating Snap, editing becomes "restricted" to the note values selected on the Snap pop-up menu (the Snap value). The Snap button and pop-up menu are located on the sequencer toolbar:

Use this pop-up menu to select the Snap value.



! Note that you can select different Snap values for the Arrange View and the Edit View.

Snap has an effect on the following operations:

→ **Moving the Song position, Locators and End marker.**

When you adjust these markers with Snap activated, they will be "magnetic" to the Snap value.

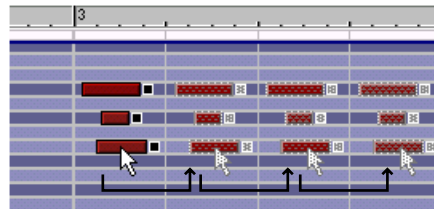
→ **Selecting events by enclosing them in a selection rectangle.**

Since the selection rectangle is magnetic to the snap value, this determines the smallest "block" you can select.

However, selecting by clicking directly on notes in the Edit View (or Groups in the Arrange View - see page 19) is not restricted by Snap.

→ **Moving and duplicating events.**

When you move one or several Events with Snap activated, they will keep their relative distance to the Snap value positions. In the example below, Snap is set to 1/4 (quarter notes):



→ **Drawing Groups in the Arrange View.**

When you create Groups with the Pencil tool, their start and end positions will be magnetic to the snap value positions. See page 18.

→ **Drawing events in the Edit view.**

The Snap value determines the smallest note position on which you can draw a note or insert a controller value or pattern change. Furthermore, the Snap value determines the smallest length of the events when you draw. See page 22.

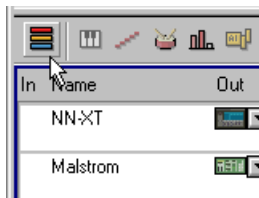
→ **Using the Eraser tool to delete events in the Edit and Arrange views.**

With Snap activated, clicking directly on events with the Eraser tool will not only delete the events "touched", but *all* events within the set snap value (e.g. 1 bar). The Eraser tool can also be used for making selection rectangles and these will be magnetic to the snap value as well. See [page 17](#).

Editing in the Arrange View

The Arrange View allows you to view several tracks at the same time, and provides a good overview of the song. This view is best suited for large-scale editing, such as rearranging blocks of music, adding or removing bars or applying quantizing and editing functions to events on different tracks at the same time.

- To select the Arrange View, click the **Arrange/Edit View** button in the top left corner of the sequencer area.

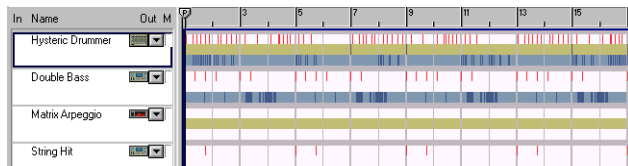


You can also toggle between Arrange View and Edit View by pressing [Shift]-[Tab] or [Command]/[Ctrl]-[E].

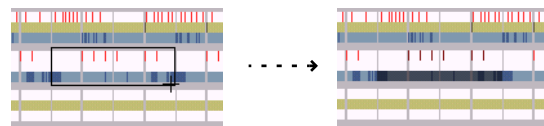
- ! On the following pages we will use the word “event” as a collective name for notes, controller changes and pattern changes.
- ! The procedures below apply to separate events in the Arrange View. Some techniques are different for Grouped events, as described on page 18.

Selecting Events

Each track in the Arrange View is divided vertically into three “lanes”, in which events are shown as thin vertical lines. The top lane shows notes (including drum notes and REX slices) in red, the middle lane shows pattern changes in yellow and the lower lane shows controller value changes in blue.



To select events in the Arrange view, click and drag a selection rectangle.



- If **Snap** is activated, the selection rectangle will be magnetic to the **Snap** value.
- You can drag a selection rectangle covering only one lane, thereby selecting only the notes, pattern changes or controllers. You can also draw a selection rectangle covering several tracks.
- If you hold down **[Shift]** when you select events, any already selected events remain selected. This allows you to make multiple, non-contiguous selections: first select some events, then press **[Shift]** and select some more events, and so on.
- You can also use the “**Select All**” function on the **Edit** menu. This selects all events, controllers and pattern changes in the song.
- The selection you make in the **Arrange View** will be maintained if you select the **Edit View**. See page 23.
- To de-select events, just click anywhere in an empty area.

Moving Events

To move the selected events, click in the selection, and drag and drop it on a new position.

- When you move the selection, it is “magnetic” to the lanes. That is, you cannot drag notes to a controller lane, etc.
- If you hold down **[Shift]** and drag, movement is restricted to horizontal or vertical only.
- If **Snap** is activated, you will only be able to drop the selection so that it maintains its relative distance to the **Snap** value positions. See page 14.

Duplicating Events

To duplicate the selected events, hold down [Option] (Mac) or [Ctrl] (Windows) and proceed as when moving events.

- **You can also use the Duplicate Track function on the Edit menu or the track context menu.**

This creates a copy of the selected track, complete with all events. The duplicated track will appear below the original track in the track list.

Using Cut, Copy and Paste

You can move or duplicate events using the Cut, Copy and Paste commands on the Edit menu. When you Paste, the events appear at the song position, on their original track(s).

- ! **If you have deleted the original tracks, or if you Paste into another Reason song document, new tracks will be created as needed.**
- **See the note below about Copying and Pasting whole tracks!**

Using Copy and Paste to repeat a section

When you Cut or Copy a selection, the song position will automatically move to the end of the selection (or, if Snap is activated, to the closest Snap value position after the end of the selection). This allows you to quickly repeat a section, in the following way:

1. **Make sure playback is stopped.**
2. **Set the Snap value to “Bar” (or to the length of the section you want to repeat, if is smaller than one bar).**
3. **Activate Snap.**
4. **Select the section you want to repeat.**

Since you can make selections over several tracks, this is a quick way to copy entire song sections.

- ! **Note that Snap doesn't apply in quite the same way when selecting Groups (see page 19). Make sure the selection contains exactly what you want before proceeding.**

5. **Select Copy from the Edit menu.**
The song position is moved to the closest snap value after the end of the selection (provided that playback is stopped).
6. **Select Paste from the Edit menu.**
The copied section is pasted in, and the song position is moved to the end of pasted section.
7. **Paste again, as many times as you want to repeat the section.**

Using Cut, Copy and Paste with Tracks

You can select one or several tracks by clicking or [Shift]-clicking in the track list. This allows you to use Cut or Copy on the track, complete with contents.

- **If you Paste the track(s) into their original song, this simply duplicates the tracks.**

However, the Pasted tracks will not be connected to any devices in the rack.

- **You can also Paste the track(s) into another song.**

Note that only the tracks (complete with contents) are copied and pasted - not their respective devices. You may want to separately copy and paste the devices to the other song.

Deleting Events

- **To delete an Event, either select it and press [Delete], [Backspace] or select Delete from the Edit menu.**

Both of these methods will delete the Event.

You can also draw selection rectangles with the Selection tool, encompassing several Events and delete them all at once. The same rules apply as when selecting Events. That is, if Snap is on, the selection rectangle will be magnetic to the Snap value positions. Also note that an Event doesn't have to be fully enclosed by the rectangle to be selected - the selection rectangle only needs to intersect or touch the Event.

Deleting Events with the Eraser tool



You can also use the Eraser tool to delete Events and Groups in the Arrange view, as well as Notes, Controller sections and Pattern Change sections in the Edit view.

The Eraser tool can be used in two ways: You can single click on events or you can make a selection rectangle encompassing several events. See below.

About Snap and the Eraser tool

With Snap activated, clicking directly on events or making selection rectangles with the Eraser tool will not only delete the events “touched”, but *all* events within the set snap value (e.g. 1 bar).

Deleting events by single clicking

- ➔ **Select the Eraser tool and click on the event you want to delete.**

When using the Eraser tool to delete events with Snap on, the following applies:

- When single clicking, all events within the set Snap value will be deleted. The “area of effect” is indicated in a dark gray color.



In this example, the Eraser tool is used to delete notes in the Edit view. Snap is set to “Bar”, and therefore a single click will remove all the instances of the note C 2 on bar 6.

Deleting events by making a selection rectangle

- ➔ **Select the Eraser tool, click and hold the mouse button and draw a selection rectangle.**

This way, you can make a selection encompassing several events and delete them all at once.

If Snap is on when a selection rectangle is drawn, it will be magnetic to the nearest snap value positions. For example, if Snap is set to “Bar”, dragging a rectangle will select all notes within an exact range of bars.



Drawing a selection rectangle with Snap set to “Bar”. All notes enclosed in the “shaded” area will be deleted.

- ! **Note that an Event doesn't have to be fully enclosed to be selected - the selection rectangle only needs to intersect or touch the Event.**

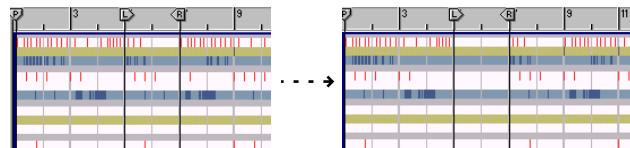
- ★ **If you hold down [Shift] when making a selection rectangle, direction is restricted to horizontal or vertical only.**

Inserting and Removing Bars

When editing the overall structure of a song, you may need to rearrange the order and length of whole sections (e.g. make the “verse” two bars shorter, add a few bars to the intro, etc.). On the Edit menu or sequencer context menu you will find two useful functions for this:

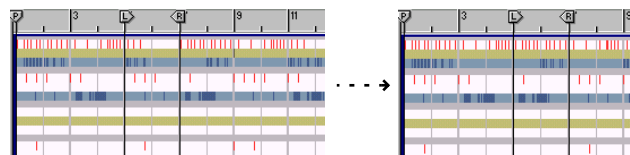
Insert Bars Between Locators

This function inserts an empty area between the locators. All events after the left locator are moved to the right to “make room” for the inserted area.



Remove Bars Between Locators

This function removes all material between the locators. All events after the right locator are moved to the left to “fill out” the gap after the removed section.



- ★ **The “Remove Bars Between Locators” function will automatically shorten any Groups intersected by the locators. This can be used as a feature in itself, as described on page 20.**

Other Editing Functions in the Arrange View

You can also apply quantizing (see page 31) and use the Change Events function (see page 32) in the Arrange View. This is useful since it allows you to edit events on several tracks in one go.

- ➔ **Note that you can select one or several tracks and have quantizing or Change Events apply to all events on the selected tracks.**
Selecting several tracks is done by [Shift]-clicking in the track list.

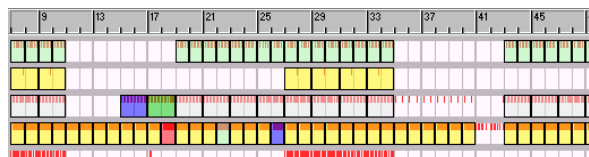
Using Groups

Sometimes it is practical to work with a section of events as one entity. This is done by Grouping the events. You may for example have a two-bar bass line that you want to move or repeat in the song - by Grouping the events, you can select, move and handle the bassline as a single object.

! This applies to the Arrange View only - you can still edit individual events in a Group in the Edit View.

Appearance and Color

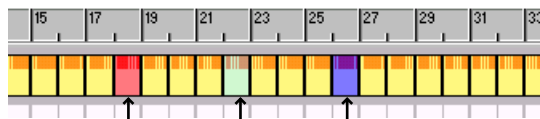
In the Arrange View, Groups appear as colored boxes.



The color of the Groups depends on their contents:

! Groups with the same color contain the same events.

This makes it easy to get an overview of the song, since variations will appear as Groups in another color.



These Groups are "variations"
- all the others have the same contents.

Creating Groups

There are two main ways to create Groups:

By using the Group command

1. Select the events that you want to Group.

It doesn't matter which lanes you select - all notes, pattern changes and controllers within the area will be included in the Group.

→ If you select events on several tracks, one Group for each track will be created.

Each Group can only contain events on one track.

2. If you want the Group to have a specific length, activate Snap and select an appropriate Snap value.

Often it is practical to create Groups that are one or several whole bars long.

3. Select Group from the Edit menu or the sequencer context menu.

Or, hold down [Command] (Mac) or [Ctrl] (Windows) and press [G].



The events are Grouped.

By drawing with the Pencil tool

1. Select the Pencil tool.

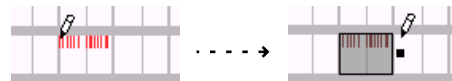
You can also select the Pencil tool momentarily by holding down [Command] (Mac) or [Alt] (Windows).

2. If you want the Group to have a specific length, activate Snap and select an appropriate Snap value.

Often it is practical to create Groups that are one or several whole bars of length.

3. Click where you want the Group to start, drag to the right and release the mouse button.

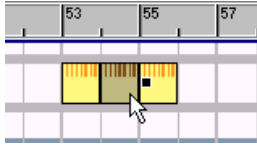
A Group is created, containing the enclosed Events. It is also possible to create empty Groups this way.



★ Groups are also automatically created when you use the "To Track", "Copy Pattern to Track" and "Convert Pattern Track to Notes" functions. See page 12.

Selecting Groups

To select a Group, just click on it in the Arrange View.



This Group is selected.

- **If you hold down [Shift] and click, you can select multiple Groups.**
You can de-select individual Groups by [Shift]-clicking them again.

- **You can also select Groups by clicking and dragging a selection rectangle, just as with events.**

If Snap is on, the selection rectangle will be magnetic to the Snap value positions. However, note that a Group doesn't have to be fully enclosed by the rectangle to be selected - the selection rectangle only needs to intersect or touch the Group.

- ⚙ **Note that it's possible to select Groups and "loose events" at the same time with this method. Make sure the selection rectangle encloses the elements you want!**

- **Another way of selecting Groups is to use the arrow keys on the computer keyboard.**

Pressing the right arrow key selects the next Group on the track, pressing the down arrow key selects the closest Group on the track below, etc. Holding down [Shift] and using the arrow keys allows you to make multiple selections.

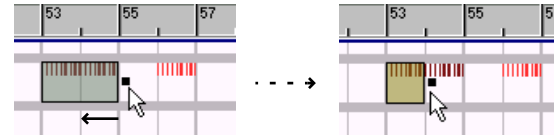
- **If you select a Group and go to the Edit View, all events in the Group will be selected.**
- **To de-select the selected Group(s), click anywhere in an empty part of the Arrange View.**

Resizing Groups

When a Group is selected, a handle appears on its right edge. You can click on this handle and drag to make the Group smaller or larger. The following rules apply:

- **If you drag to the left to make the Group smaller, any events that end up outside the Group boundary are no longer included in the Group.**

As a consequence, if you drag the handle all the way past the start of the Group, all events are Ungrouped (see below).

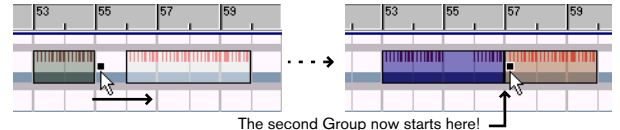


- **If you drag to the right to make the Group larger, any events you enclose will become part of the Group.**



- **Note: Groups cannot overlap!**

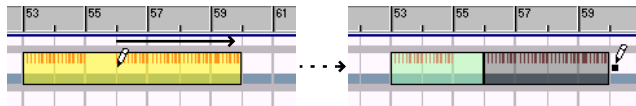
This means that if you enlarge a Group so that it partially covers another Group, this will automatically resize the other Group as well:



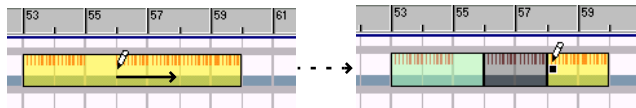
The second Group now starts here!

Dividing Groups

You can divide a Group into two by clicking with the Pencil tool at the desired position and dragging to the end of the Group.



Actually, this is just a consequence of the fact that Groups cannot overlap. As soon as you create a Group that overlaps another Group, the other Group is automatically resized. For example, if you were to draw a small Group within a larger Group, you would end up with three Groups:



Tip: Splitting Groups on several Tracks

If you have Groups on several tracks and want to split these at the same position, you can use the following method:

1. **Set both the Left and the Right Locator to the desired split position.**
2. **Select “Insert Bars Between Locators” from the Edit menu.**
The Groups are split.

Combining Groups

There are two main ways to combine two or more Groups into one:

By using the Group command

1. **Select the first and the last Group that you want to combine.**
All Groups in between these will be included as well.
2. **Select Group from the Edit menu.**
You will now have one larger Group.

By Resizing

1. **Click the size handle of the first Group and drag to the right.**
2. **Release the mouse button at the end of the last Group.**
All Groups in between are combined to one larger Group.

Find Identical Groups

This command on the Edit menu helps you locate all Groups with the same contents:

1. **Select a Group.**
2. **Select “Find Identical Groups” from the Edit menu.**
All Groups with the same contents are selected in the Arrange View.

Ungrouping

There are two ways to dissolve a Group:

- **Select it and select Ungroup from the Edit menu or sequencer context menu,**
- or
- **Click on the Group size handle and drag it all the way to the left.**

Neither of these methods affect the events in the Group, they just remove the Grouping.

Editing with Groups

You can work with Groups much like you edit selected events in the Arrange view:

- **To move a Group, click on it and drag it to a new position, taking the Snap value into account.**
If you move the Group so that it partially overlaps another Group, the other Group will automatically be resized. If the moved Group overlaps the other Group completely, you will get one large Group containing the events from both.
- **To duplicate a Group, hold down [Option] (Mac) or [Ctrl] (Windows) and proceed as when moving.**
This duplicates the Group and all its contents. You can also use Copy and Paste for this, following the same rules as for selected events.
- **To delete a Group, either select it and press [Delete], [Backspace] or select Delete from the Edit menu.**

or

- **Select the Eraser tool and click on a Group.**

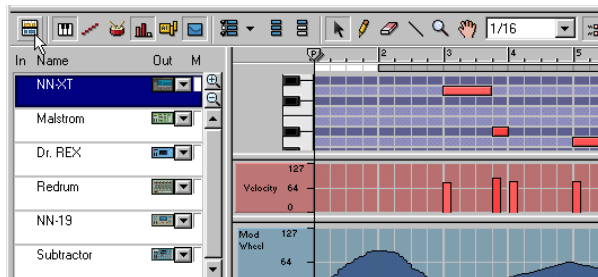
Both of these methods will delete the Group and all its contents.

You can also draw selection rectangles with the Selection tool or the Eraser tool, encompassing several groups and delete them all at once. The same rules apply as when selecting groups. That is, if Snap is on, the selection rectangle will be magnetic to the Snap value positions. Also note that a Group doesn't have to be fully enclosed by the rectangle to be selected - the selection rectangle only needs to intersect or touch the Group.

The Edit View

The Edit View allows you to perform detailed editing to the events on a single track. This is also where you create notes, pattern changes and controller values from scratch by drawing.

- To select the Edit View, click the Edit/Arrange View button in the top left corner of the sequencer area.



You can also toggle between Arrange View and Edit View by pressing [Shift]-[Tab] or [Command]/[Ctrl]-[E].

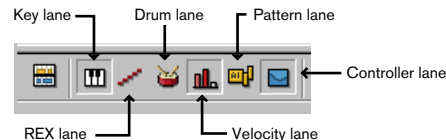
Selecting a Track for Editing

The Edit View shows the events of the track that has the *focus* in the track list.

- If one track is selected when you enter Edit View, that track will have the focus and its events will be shown.
- If more than one track is selected in the track list when you select Edit View, the track you last clicked on will have focus.
- You can change edit track at any time, by clicking in the track list. This way you can stay in the Edit View and select different tracks for editing, without having to go back to the Arrange View.

About the Lanes

The Edit View is (or can be) divided vertically into lanes. There are six different lanes, suitable for editing different types of events. Any combination of lanes can be shown. You show and hide lanes by clicking their respective buttons in the sequencer toolbar:



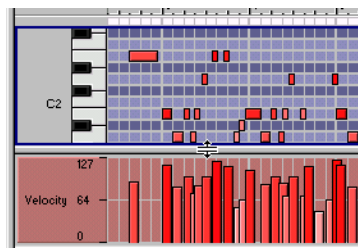
- If you hold down [Option] (Mac) or [Alt] (Windows) and click a Lane button, only that lane will be shown (all other lanes are hidden).

By default, the lanes that are shown when you select Edit View depends on the device type to which the track is connected (and whether the track contains controller data). For Redrum tracks, the Drum lane, Velocity lane and Pattern lane are shown, for Dr.Rex tracks, the REX lane and Velocity lane are shown, and so on.

However, once you show or hide lanes, the new combination of lanes will be stored individually for each track. The next time you select Edit View for that track, the lane configuration will be the same.

Resizing and Zooming

- You can resize lanes by dragging the dividers between them.



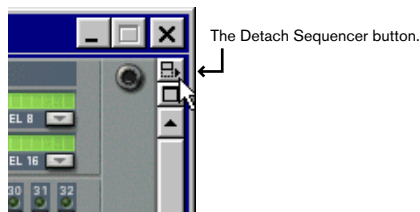
- Where applicable, the lanes have individual zoom controls and scrollbars.
- The Magnifying Glass tool can be used for zooming in and out. Click to zoom in, and click while pressing [Option] (Mac)/[Ctrl] (Windows) to zoom out.

→ **The Hand tool can be used for scrolling the view.**

Just click, hold and drag in the desired direction.

→ **For extensive editing, you may want to detach the sequencer area from the rack and use it in a separate window.**

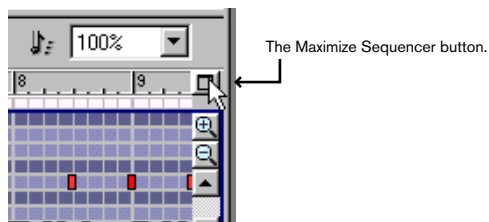
This is done either by clicking the Detach Sequencer button in the rack or by selecting "Detach Sequencer Window" from the Windows menu.



To reattach the sequencer, either click the Attach Sequencer button (in the rack or in the detached sequencer window) or select "Attach Sequencer Window" from the Windows menu.

→ **Alternatively, you can also maximize the sequencer area so that it fills the rack.**

This is done by clicking the Maximize Sequencer button or by holding down [Command] (Mac) or [Ctrl] (Windows) and pressing [2] on the left part of the computer keyboard.



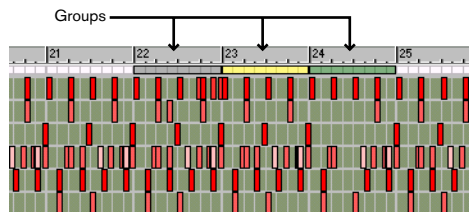
About the Ruler and the Group strip

At the top of the Edit View you will find the ruler. Just like the ruler in the Arrange View, this shows meter positions (bars and beats), helping you find the right positions in the song.

→ **You can adjust the horizontal zoom individually for the Edit View and the Arrange View.**

This makes sense, as you will probably work with a larger magnification when performing fine editing.

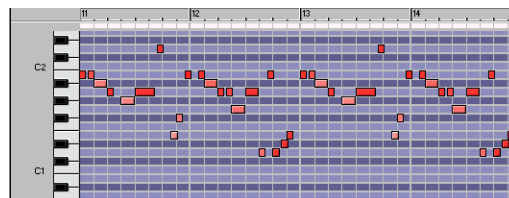
Just below the ruler is a narrow empty strip. This shows the Groups (if any) as colored bars, providing additional means of orientation in the Edit view.



★ When you edit events within a Group, you will note that the Group indicator changes color. This is because the color of a Group depends on its contents, as described on page 18.

Drawing and Editing Notes

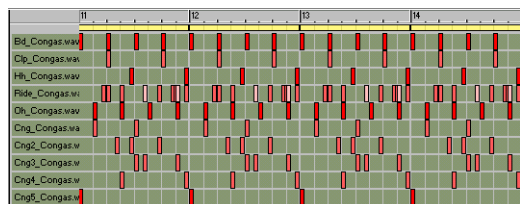
Notes are drawn and edited in one of three lanes: the Key lane, the Drum lane and the REX lane:



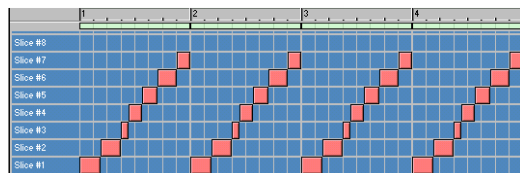
The Key lane. The piano keyboard to the left indicates the pitch of the notes, covering the whole MIDI note range (C-2 to G8). Note that the black and white keys are reflected in the

background colors of the grid, making it easier to find the right pitch when drawing and moving notes!

This is the lane to use when editing Synth or Sampler tracks.



The Drum lane. This is divided vertically into ten pitches, corresponding to the ten drum sound channels on a Redrum device (and named accordingly, if the track is connected to a Redrum device). Use this for editing drum tracks.



The REX lane. This is divided vertically into pitches (from C3 and up), corresponding to the slices in a Dr.Rex loop player device. Use this for editing Dr.Rex tracks.

- ➔ **In all three lanes, the actual notes are shown as “boxes”, with the note length indicated by the width of the box and the velocity values indicated by the color of the box (the darker the color, the higher the velocity).**

The basic note editing procedures are the same for all three lanes.

Drawing notes

- If you want to restrict note input to certain note values (e.g. sixteenth notes), set the Snap value accordingly and activate Snap.**
- Select the Pencil tool.**
You can toggle temporarily between the Arrow tool and the Pencil tool by holding down [Command] (Mac) or [Alt] (Windows).
- If needed, click in the piano keyboard display, drum sound list or slice list to find the correct pitch.**
If the track is connected to a device, this will play the corresponding note.
- Click in the note display part of the lane, at the desired position.**
A note will be inserted at the closest Snap value position.

- ➔ **If you just click, the note will get the length of the Snap value.**
This is true regardless of whether Snap is activated or not.
- ➔ **If you instead click and keep the mouse button pressed, you can drag to the right to set the length of the note.**
If Snap is on, the length will be a multiple of the Snap value (unless you hold down [Shift] while you drag). Also, see the note about drum note lengths below.

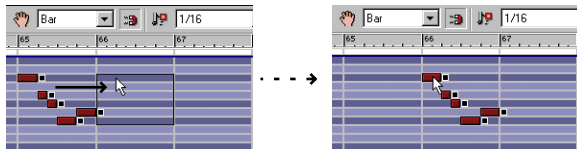
Selecting notes

To select notes in the Edit View, use one of the following methods:

- ➔ **Click on a note with the Arrow tool to select it.**
- ➔ **To select several notes, hold down [Shift] and click.**
You can de-select individual notes by [Shift]-clicking them again.
- ➔ **You can also click and drag a selection rectangle around the notes you want to select.**
If Snap is on, the selection rectangle will be magnetic to the nearest snap value positions. For example, if Snap is set to “Bar”, dragging a rectangle will select all notes within an exact range of bars (and within the pitches enclosed by the rectangle).
- ➔ **You can select the next or previous note on the track by pressing the right or left arrow key on the computer keyboard.**
Holding down [Shift] and using the arrow keys allows you to make multiple selections.
- ➔ **To select all notes on the track, use the Select All function on the Edit menu.**
Make sure that the correct lane (Key, Drum or REX) has focus first - otherwise you may select all controllers or pattern changes. To set focus to a lane, click somewhere in it (focus is indicated by a thin extra border within the lane).
- ➔ **To deselect all notes, click somewhere in an empty area.**

Moving notes

- **To move a note, click and drag it to a new position.**
If several notes are selected, all will be moved. The individual distance between the moved notes will be kept.
- **If Snap is on, the moved events will keep their relative distance to the Snap value positions.**
For example, if Snap is set to "Bar", you can move the selected notes to another bar without affecting their timing.



- **If you hold down [Shift] when you drag, movement is restricted to horizontal or vertical only.**
This helps you move notes without accidentally transposing them, or transposing notes without accidentally changing their meter position.

Duplicating notes

To duplicate the selected notes, hold down [Option] (Mac) or [Ctrl] (Windows) and proceed as when moving notes.

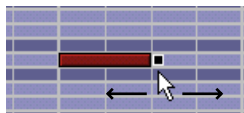
Using Cut, Copy and Paste

You can move or duplicate events using the Cut, Copy and Paste commands on the Edit menu.

- **When you Cut or Copy, the song position is automatically moved to the end of the selection (or, if Snap is activated, to the closest Snap value position after the end of the selection).**
You can use this for repeating events, as described on page 16.
- **When you Paste, the events appear at the song position, on their original track(s).**

Resizing notes

When you select a note, a handle appears on its right edge. You can click on this handle and drag to make the note shorter or longer.



- **If Snap is on, the end of the note will be magnetic to the Snap value positions.**
You can disable this function temporarily by pressing [Shift] when you drag. This allows you to resize the note to any length, regardless of the Snap value.
- **If several notes are selected, all will be resized by the same amount.**

About resizing drum notes

Drum notes can be resized as any other notes. However, the result of this depends on the settings of the Decay/Gate switch and the Length knob for the drum sound on the Redrum panel:

- **If Decay mode is selected, the drum sound will play to its end, regardless of the note length.**
Or rather, it will fade out according to the Length setting.
- **If Gate mode is selected, the note length affects the resulting sound.**
However, the maximum length of the sound is set by the Length knob - the sound will be cut off after this length, regardless of the note length. Finally, even if the Length knob is set to its maximum value, the sound will not play longer than the length of the drum sample.

Deleting notes

You can delete notes in two ways:

- **Select them and press [Backspace] or [Delete], or select Delete from the Edit menu.**
- **Select the Eraser tool and click on the notes you want to delete.**
You can also drag a selection rectangle with the Eraser tool and delete all notes encompassed by the rectangle.

When using the Eraser tool with Snap on, the following applies:

- When single clicking, all notes of the same pitch within the set Snap value will be deleted. The “area of effect” is indicated in a dark gray color.

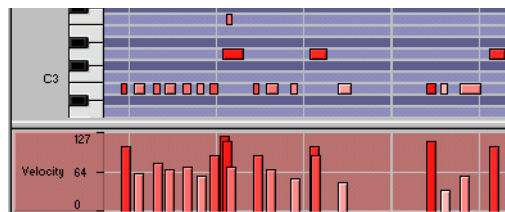


In this case, with Snap set to “Bar”, a single click will remove all the instances of the note C 2 on bar 6.

- If a selection rectangle is drawn, it will be magnetic to the nearest snap value positions. For example, if Snap is set to “Bar”, dragging a rectangle will select all notes within an exact range of bars.
- **If you hold down [Shift] when making a selection rectangle, direction is restricted to horizontal or vertical only.**

Editing velocity

The velocity values of notes are edited in the Velocity lane.



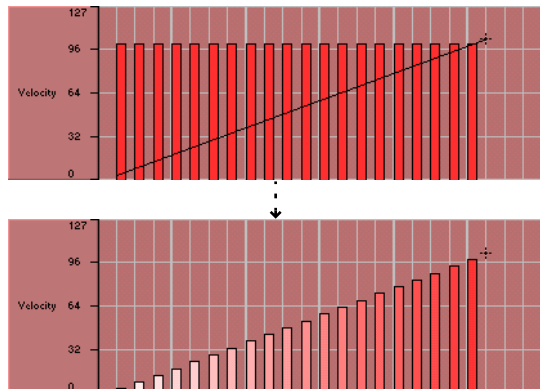
The velocity values are shown as bars, with higher bars indicating higher velocity. Note also that the color of the notes and bars reflect the velocity.

To change the velocity of a note, click on its velocity bar with the Pencil tool and drag the bar up or down. Clicking above a bar immediately raises the velocity to the level at which you click.

Creating velocity ramps and curves

You can also edit the velocity of several notes at once, in two ways:

- **By dragging the Line tool across the bars, at the desired height.**



Drawing a velocity ramp with the Line tool.

- **By dragging the pencil across the bars, at the desired height.**

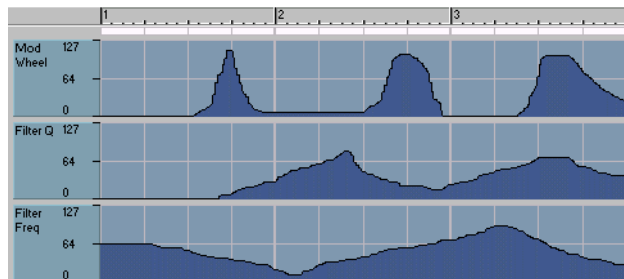
The Line tool is probably the preferred method for creating regular, smooth ramps, or for giving all the notes the same velocity (by drawing a straight line), while the Pencil tool can be used for creating more irregular curves.

! If you hold down [Shift] when you edit velocity values, only the selected notes will be affected!

This can be very useful, especially in “crowded” sections with lots of notes. Consider for example if you have a busy drum beat, and want to adjust the velocity of the hi-hat notes only. Simply dragging with the line- or pencil tool would change the velocity of all other drum notes in the area too, but if you first select the hi-hat notes in the Drum lane and press [Shift] as you draw, you can edit their velocity without affecting any other notes!

Editing Controllers

Controllers are shown and edited in the Controller lane. This lane in turn is divided into several “subtracks”, one for each automatable parameter for the corresponding device.



The Controller lane for a Subtractor track, with three controllers shown.

Showing and Hiding Controllers

For each track, you can select which controllers should be shown. This can be done in several ways:

- **Hold down [Option] (Mac) or [Alt] (Windows) and click on a parameter on a device panel in the rack.**

This sets focus to the first sequencer track connected to the device, opens Edit View, brings the Controller lane and shows the automation subtrack for the specified parameter, all in one go.

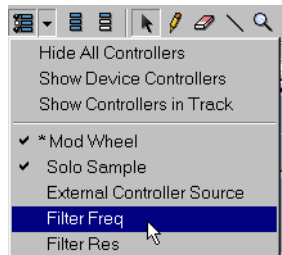
- **You can do the same thing by selecting “Edit Automation” on the context menu for the parameter.**

You bring up the parameter context menu by [Ctrl]-clicking (Mac) or right-clicking (Windows) on the parameter on the device panel.

- **If you use a Mac with a two-button mouse, it's a good idea to assign [Ctrl]-click to the right mouse button, allowing you to bring up context menus by right clicking.**

- **By using the Controller pop-up menu on the sequencer toolbar, you can hide or show individual controllers from the sequencer.**

Shown controllers are indicated by a tick mark on the pop-up menu - select a controller to show it or hide it. Controllers for which there is data (automation) in the track are indicated with an asterisk next to the controller name.



- **Click the “Show Device Controllers” button to show all controllers available for the track’s device.**



- **Click the “Show Controllers in Track” button to show all controllers for which you have recorded or drawn automation in the Track.**

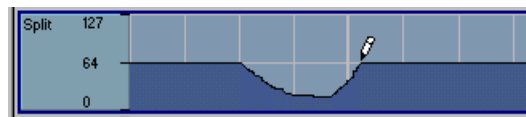


- **Select “Hide All Controllers” from the Controller pop-up menu to hide all controllers.**

This will leave the Controller lane empty.

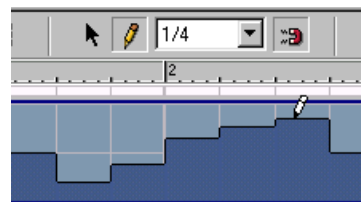
Drawing and Editing Controllers

Regardless of whether you’re editing recorded controllers or creating controller changes from scratch, you do it by drawing with the Line- or Pencil tool.



Note:

- **When using the line tool, you can hold down [Shift] when drawing to restrict movement to horizontal only.**
- **If Snap is on, the controller value change you enter will snap to the nearest Snap value position.**
Also, the length of the changed section will be a multiple of the Snap value.

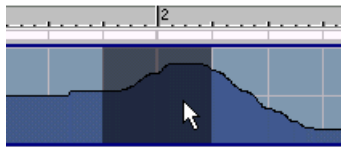


In this example, Snap is set to 1/4. Thus, the controller changes you enter will be in “steps”, one or more quarter notes in length.

- **If the controller hasn’t been automated yet (the words “Not Automated” are shown in the Controller lane), it is a good idea to first set the parameter to a good “default value” on the device panel.**
The reason is that as soon as you enter a controller value, the rest of the track will be filled with the original value of the parameter (the value set on the device panel). This works exactly the same as when recording controllers - see page 8.

Selecting sections of a controller track

To select a section of the “subtrack” for a controller, click and drag a selection rectangle with the Arrow tool. If Snap is on, the selection will be magnetic to the Snap value positions, just as when selecting notes.



The selected section is shown as a shaded rectangle.

- **By holding down [Shift] and dragging, you can select multiple, discontinuous sections of the controller subtrack.**

⚙️ If you select Groups or sections of the controller lane in the Arrange View, this section will be selected when you got to Edit View and vice versa.

Moving and Duplicating Controller sections

- **To move a selected controller section, click and drag it to another position on the same subtrack.**
Snap is taken into account as usual.
- **To duplicate a selected controller section, hold down [Option] (Mac) or [Ctrl] (Windows), click and drag it.**
- ! **Moving or duplicating controllers will replace the controller values at the new position (just as if you had edited them with the Line- or Pencil tool).**

Using Cut, Copy and Paste

You can move or duplicate selected controller sections using the Cut, Copy and Paste commands on the Edit menu.

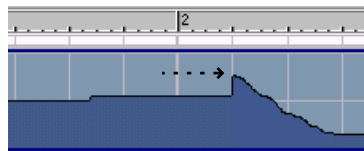
- **When you Cut or Copy, the song position is automatically moved to the end of the selection (or, if Snap is activated, to the closest Snap value position after the end of the selection).**
You can use this for repeating events, as described on page 16.
- **When you Paste, the controller section appears at the song position, on its original subtrack.**

Deleting Controller sections

You can delete controller sections in two ways:

- **By making a selection (as described above) and pressing [Backspace] or [Delete] or by selecting Delete from the Edit menu.**
- **By using the Eraser tool.**
If Snap is on, you can single click to immediately erase the shaded area which corresponds to the set Snap value (e.g. Bar). You can also make a selection range by clicking and dragging.

The result is this:



The controller value just before the deleted selection will remain until the end of the selection.

- ! **You can't remove all automation using this method - there will always be at least one controller value left. To remove all automation, use the Clear Automation function:**

Clearing Automation

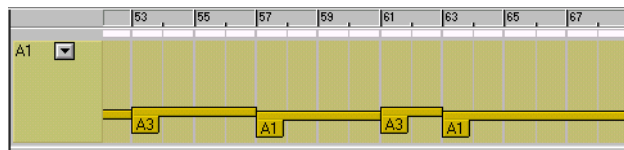
To remove all automation for a controller, select “Clear Automation” from one of the following menus:

- **The context menu for the controller subtrack.**
This appears when you [Ctrl]-click (Mac) or right-click (Windows) in the subtrack.
- **The Edit menu.**
Requires that the controller subtrack has focus. Click in the subtrack if you are uncertain.
- **The parameter context menu.**
This appears when you [Ctrl]-click (Mac) or right-click (Windows) on the parameter on the device panel.
Note that this clears all automation for the parameter, on all tracks!

Selecting “Clear Automation” will remove all controller values from the subtrack, and the text “Not Automated” will be shown.

Inserting and Editing Pattern Changes

Pattern changes are viewed and edited in the Pattern lane:



A pattern change is shown as a yellow “tab” with the Bank and Pattern number. From the tab, a bar stretches to the right, for as long as the selected pattern is “active”, i.e. to the next pattern change.

! **When you record pattern changes, they are automatically positioned on downbeats (at the beginning of new bars).**

Inserting Pattern Changes

To insert a Pattern change, proceed as follows:

1. **If you haven’t automated any pattern changes for the track yet (the words “Not Automated” are shown in the Pattern lane), it is a good idea to first select a “default pattern” in the pattern device.**

This is especially useful if you are using a main pattern and want to insert changes to variation pattern here and there.

The reason is that just like when you record pattern changes, the rest of the track will be “filled” with the original value as soon as you enter a pattern change somewhere on the track.

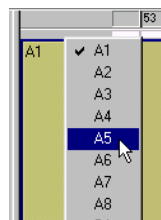
2. **Activate Snap and set the Snap value to the note position where you want to insert pattern changes.**

It is probably a good idea to set Snap to “Bar”, at least if you are working with patterns of a length corresponding to the time signature (e.g. 16 or 32 step patterns and 4/4 time signature). However, if you are working with patterns of another length, it can make sense to use other Snap values.

- ! **Don’t insert pattern changes with Snap turned off, unless you want chaotic rhythm changes!**

3. **Pull down the Pattern pop-up menu to the left in the Pattern lane, and select the pattern you want to insert.**

The selected pattern is shown next to the pop-up menu.

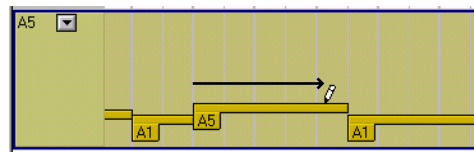


The patterns are listed with the bank letter and pattern number (A1, A2, A3, and so on).

4. **Click with the Pencil tool at the position where you want the pattern change to happen, and keep the mouse button pressed.**

5. **Drag to the right.**

When you drag, you will see the previous or original pattern being replaced by the pattern you insert.

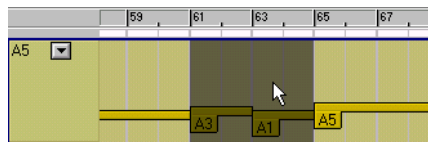


6. **Release the mouse button at the position where you want the pattern change to “end”.**

! **The “Pattern Enable/Mute” switch (the button above the pattern selection buttons on the device panel, used for temporarily silencing the pattern playback) is automated using controller automation. The controller is called “Pattern Enabled”.**

Selecting Pattern Changes

To select a section of Pattern lane, click and drag a selection rectangle with the Arrow tool. If Snap is on, the selection will be magnetic to the Snap value positions, just as when selecting notes.



The selected section is shown as a shaded rectangle.

→ **By holding down [Shift] and dragging, you can select multiple, discontinuous sections of the Pattern lane.**

★ **If you select Groups or sections of the pattern lane in the Arrange View, this section will be selected when you got to Edit View.**

Moving and Duplicating Pattern Change sections

You can move and duplicate selected sections of the Pattern lane, just as when moving controller sections. Just like when inserting pattern changes, it is recommended that Snap is activated (and in most cases set to "Bar") when you do this.

You can also move or duplicate sections using the Cut, Copy and Paste commands on the Edit menu. Again, the same rules apply as when editing controllers.

Deleting Pattern Change sections

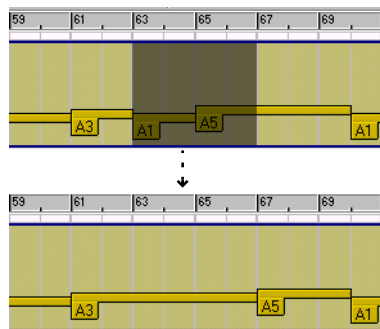
You can delete a section of the Pattern Lane in two ways:

→ **By making a selection (as described above) and pressing [Backspace] or [Delete] or by selecting Delete from the Edit menu.**

→ **By using the Eraser tool.**

If Snap is on, you can single click to immediately erase the shaded area which corresponds to the set Snap value (e.g. Bar). You can also make a selection range by clicking and dragging.

The result is this:



The pattern before the deleted section will remain selected until the end of the section.

! **Again, make sure Snap is activated.**

! **You can't remove *all* pattern change data using this method. To remove *all* pattern automation, use the Clear Automation function:**

Clearing Automation

To remove all pattern changes, proceed as follows:

1. **[Ctrl]-click (Mac) or right-click (Windows) in the Pattern lane.**
The context menu appears.

2. **Select "Clear Automation".**

This will remove all pattern changes from the track, and the text "Not Automated" will be shown.

Quantizing

The Quantize function moves recorded notes to (or closer to) exact note value positions. This can be used for correcting errors, “tightening up” recorded music or changing the rhythmic feel.

Applying Quantizing

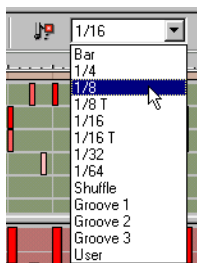
In Reason, you use the Quantize function in the following way:

1. Select the notes you want to quantize.

Only notes will be affected, so you can select Groups or complete Tracks if you like.

2. Pull down the Quantize pop-up menu on the sequencer toolbar and select a Quantize value.

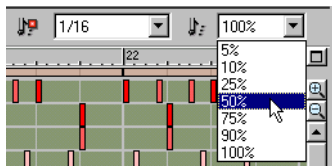
This determines to which note values the notes will be moved when you quantize. For example, if you select sixteenth notes, all notes will be moved to (or closer to) the closest sixteenth note position.



The Quantize pop-up menu.

3. Select a value from the Quantize Strength pop-up menu.

This is a percentage, governing how much each note should be moved. If you select 100%, notes will be moved all the way to the closest Quantize value positions; if you select 50%, notes will be moved half-way, etc.

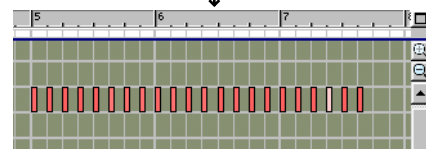


4. Click the Quantize button or select “Quantize Notes” from the Edit menu.

The selected notes are quantized.



The Quantize button



In this example, a sloppily recorded hi-hat pattern is quantized to straight eighth notes (Quantize value 1/8, Strength 100%).

Quantizing to Shuffle

On the Quantize pop-up menu, you will also find an option called “Shuffle”. If this is selected when you quantize, the notes are moved towards sixteenth note positions, but with the Shuffle applied.

As described in the Getting Started book, Shuffle creates a “swing feel” by delaying the even-numbered sixteenth notes (the sixteenth notes that fall in between the eighth notes). The amount of Shuffle is set with the Pattern Shuffle control on the transport panel.



The Pattern Shuffle control.

Quantizing to Shuffle is useful if you want to match the timing of recorded notes with pattern devices in the song (if Shuffle is activated in the patterns).

→ **The Quantize Strength setting applies as when quantizing to regular Quantize values.**

Quantizing to Grooves

The Quantize pop-up menu also contains three items named “Groove 1-3”. These are three different, slightly irregular rhythmic patterns. If you select one of these as Quantize value and apply Quantize, your notes will be moved towards the note positions in the Groove pattern, creating different rhythmic feels.

Creating your own Groove

You can create your own groove and apply this using Groove Quantize:

1. **Create or record a rhythmic note “pattern” of some kind.**
You may for example record a drum pattern, or use the notes playing the slices in a REX loop.
 2. **Select the notes you want to include in the user groove.**
The groove can be of any length, but it's usually most practical to make it one or two bars long.
 3. **Select “Get User Groove” from the Edit menu or sequencer context menu.**
Your pattern is stored as the User groove.
 4. **Select any notes that you want to quantize, make sure “User” is selected as Quantize value, and quantize as usual.**
The rhythmic feel of your groove is applied to the notes.
- ! **The User Groove is only stored temporarily - it isn't included when you save your Song.**

Quantizing during recording

You can have Reason quantize notes automatically when they are recorded. This is done by activating the “Quantize Notes during Recording” button on the sequencer toolbar, before you start recording.

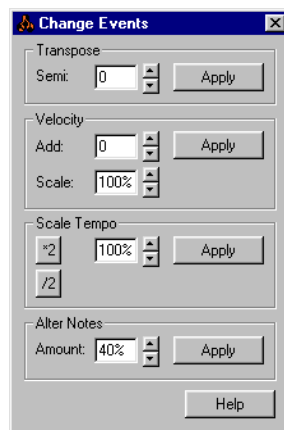


The Quantize value and Strength settings apply as usual.

The Change Events Dialog

The Change Events dialog contains some special editing functions. Proceed as follows:

1. **Select the events to which you want to apply the editing functions (in the Arrange view or Edit view).**
The Change Events functions are mainly used with notes, but the Scale Tempo function will also affect controllers and pattern changes (see below).
2. **Select Change Events from the Edit menu or the context menu for the selected events.**
The Change Events dialog appears.



3. **Make settings for one of the functions in the dialog and click the Apply button next to the settings.**
All settings can be made by clicking the spin controls or by clicking in a value field and entering a value numerically. The functions are described below.
4. **If you like, use other settings in the same way.**
You can use the transport controls as usual while the dialog is open. This allows you to play back the events to check out the changes.
5. **When you are done, close the dialog.**

Transpose

This function transposes the selected notes up or down, by the specified number of semitones.

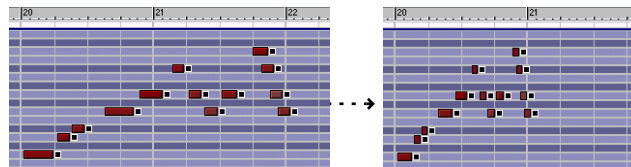
Velocity

Adjusts the velocity of the selected notes.

- **The Add field lets you add a fixed amount to the velocity values.**
To subtract, enter a negative amount. Note that the possible velocity range is 1-127. Adding an amount to a note with velocity 127 will not make any difference.
- **The Scale field allows you to scale velocities by a percentage factor.**
Scaling with a factor above 100% will increase the velocity values, but also make the difference between soft and hard notes bigger. Scaling with a factor below 100% will decrease the velocity values, but also make the difference between soft and hard notes smaller.
- **By combining the Add and Scale functions, you can adjust the “dynamics” of the notes in various ways.**
For example, by using a Scale factor below 100% and Add a suitable amount, you can “compress” the velocity values (decreasing the difference between the velocity values without lowering the average velocity).

Scale Tempo

This function will make the selected events play back faster (Scale factor above 100%) or slower (Scale factor below 100%). This is achieved by changing the position of the events (starting from the first selected event) and adjusting the length of the notes accordingly.



The result of applying Scale Tempo with the Scale factor 200% (double speed).

- **The buttons [2] and [/2] are “shortcuts” to Scale factors 200% and 50%, respectively.**
These are probably the most common values used, simulating double tempo and half tempo.
- ! **This function affects all types of events: notes, controllers and pattern changes!**

Alter Notes

This function alters the properties pitch, length and velocity of the selected notes, in a random fashion.

- **The function will only “use” values that already exist among the selected notes.**
For example, if you have selected notes within a specific pitch interval, the altered notes will remain within this pitch interval. Similarly, only velocity values and note lengths that were already used in the selection will be applied by the Alter function. You could say that the function “shuffles” the existing properties in a selection and redistributes them among the notes.
- ! **This means that the less variation there is among the selected notes, the less the effect of the Alter function.**
- **You can adjust the amount of Alteration with the Amount value.**
- ⚙ **This function is especially useful for experimenting with REX loops.**
Select some notes on a Dr.Rex track and use Alter Notes to create instant variations, without losing the timing and rhythmic feel of the loop!

Importing and Exporting MIDI Files

Reason can import and export standard midi files (SMF). This allows you to transfer MIDI data between Reason and other applications.

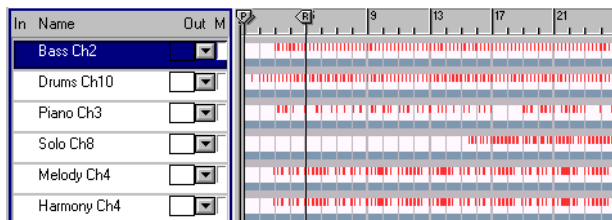
Importing a MIDI File

To import a Standard MIDI File, select “Import MIDI File” from the File menu. In the file dialog that appears, locate and open the MIDI file.

→ Under Windows, MIDI files have the extension “.mid”.

On a Macintosh, MIDI files are recognized if they have the file type “Midi”.

Now, a number of new tracks are created in Reason’s sequencer. The tracks will have their original name, with their original MIDI channel added.



→ If the imported MIDI file is of “Type 1”, there will be one sequencer track for each track in the MIDI file.

→ If the imported MIDI file is of “Type 0” (that is, it contains one track with MIDI events on multiple channels), there will be one sequencer track for each used MIDI channel.

→ Any tempo changes in the MIDI file are disregarded.

The tempo in Reason will be set to the first tempo in the MIDI file.

→ The new tracks will not be connected to devices in the rack.

You will need to connect the tracks manually to the proper devices, by using the Out pop-up menu in the track list.

→ All controller data in the MIDI file is included.

This means that pitch bend, volume and modulation wheel data are preserved properly. However, some controllers may “mean” different things for the original MIDI instruments used when creating the MIDI file and the devices in Reason. When you have connected a sequencer track to a device, you may therefore need to remove some unwanted automation from the track.



Green frames will appear for the automated parameters in the device panels. This helps you locate any unwanted controller data.

Exporting a MIDI File

To export your Reason song as a MIDI file, proceed as follows:

1. Set the End (E) marker at where you want the MIDI file to end.

The MIDI file will contain all events on all tracks from the start of the song to the End marker.

2. Select “Export MIDI File” from the File menu.

3. In the file dialog that appears, specify a name and location for the file.

Under Windows, the file will automatically get the extension “.mid”. Under Mac OS, this is not required. However, if you want the MIDI file to be recognizable under Windows (and by some hardware sequencers), you may want to activate the option “Add Extension to File Name” before saving.

4. Click Save.

MIDI files exported by Reason will have the following properties:

→ The MIDI file will be of Type 1, with one MIDI track for each track in the Reason sequencer.

The tracks will have the same names as in the Reason sequencer.

→ Since the Reason sequencer doesn’t use MIDI channels as such, all tracks will be set to MIDI channel 1.

→ The sequencer tempo is included in the MIDI file.



REASON

2

→ Routing Audio and CV

About the various signals that can be routed

This chapter describes the various ways you can route signals in Reason. The following signal types are used:

Audio

Apart from the Matrix Pattern Sequencer, all devices have audio connectors on the back. The audio connectors carries audio signals to or from devices via virtual “cables”.

- **Audio connectors are shown as large “quarter inch” jacks.**
- **Audio Effects devices, which are used to process audio, have both audio inputs and outputs.**
- **Instrument devices, which generate audio, have either mono or stereo left/right audio output connectors.**
You do not have to use both outputs for devices with stereo outputs. Use the left output to get a mono signal from a stereo device.
- **To monitor audio outputs from devices, the signals can be either be routed via a mixer - or directly- to the physical outputs of your audio hardware.**
Typically, if you are using audio hardware with standard stereo outputs, you will most probably use one or several mixers in Reason to mix the audio signals to the master outputs.

CV/Gate

CV (control voltage) signals are used to modulate parameter values, and do not carry audio. Gate signals are also a type of control voltage, but are “normally” used for slightly different purposes.

- **CV/Gate connectors are shown as smaller “mini” jacks.**
- **CV is typically used for modulation purposes.**
For example you could modulate one parameter with the value produced by another parameter.
- **Gate outputs/inputs are typically used to *trigger* events, such as note on/off values, envelopes etc.**
Gate signals produce on/off values, plus a “value” which could be likened to (and used as) velocity.
- **You can only route CV/Gate signals from an output to an input (or vice versa).**
You cannot route an input to another input or an output to another output.

MIDI Routing

There are several ways you can route MIDI from external MIDI devices to Reason devices. This is described in the chapter “Routing MIDI to Reason”.

About Cables

Hiding and Showing

If you have made many connections in Reason, the cables can sometimes obscure the view, making it difficult to read the text printed on the back panels of the devices. You can hide all cables in the following way:

- **To hide all cables, press [Command]+[L] (Mac) or [Ctrl]+[L] (Windows), or (de)select “Show cables” on the Options menu.**
When cables are hidden, connections are indicated by a colored connector. Repeating the above procedure make the cables appear again.



Cables hidden

- **When hidden, you can still connect or disconnect cables in the same way as when they are shown.**
See [page 38](#) for a description of the available routing methods.

Checking Connections

It is possible to check to which device a jack is connected (useful if the cables are hidden, or if the connected devices are located far apart in the rack):

- **Positioning the pointer over a connector.**
A tool tip appears after a moment, showing the device and the specific connector at the other end.

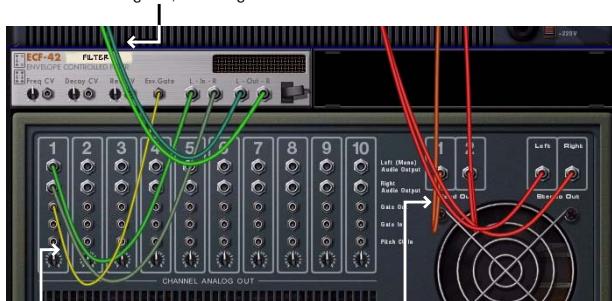


Color Coding

Cables are color coded in the following way, making it easier to discern between the various connections:

- Audio connections are different shades of red.
- CV connections are different shades of yellow.
- Connections to and from Effects devices are different shades of green.

These cables are green, indicating effect device connections.



This cable is yellow, indicating a CV connection.

These cables are red, indicating connections between instruments and mixer devices.

Automatic Routing

Auto-routing is when devices are automatically routed according to default rules. Auto-routing is performed in the following circumstances:

- When a new device is created.
- When moving, duplicating or pasting devices with [Shift] pressed.

Automatic Routing Rules

Reason Mixer Device

- ➔ **The first created mixer device will be routed to the first available input pair in the Hardware Device.**
If more mixers are created they will be connected via the mixers Chaining connectors (see the Mixer chapter).

Routing Devices to the Mixer

- ➔ **When an Instrument Device is created, it is auto-routed to the first available mixer channel(s).**

Routing a Send Effect to the Mixer

- ➔ **When you have a mixer selected and create an effect device, it will be connected as a send effect (to the first free Aux Send/Return).**
Examples of effects that lend themselves well for use as send effects are reverb, delay and chorus.

Routing an Effect Directly to a Device (Insert)

- ➔ **When you have an instrument device selected and create an effect, that effect will be connected as an insert effect. That is, the signal from the device will pass through that effect and to the mixer (or to another effect).**

CV/Gate Auto-route

- ➔ **The only instance of CV/Gate auto-routing in Reason is when you create a Matrix Pattern Sequencer with either a Subtractor or NN-19 Sampler selected.**
The Matrix Note and Gate CV outputs are automatically connected to the Sequencer Control CV and Gate inputs on the instrument device, respectively.

Auto-routing Devices after they have been Created

Here follows some additional rules about auto-routing devices that are already in the rack:

- ➔ **To reroute a device already in the rack, you can select it and use Disconnect Device and Auto-route Device, both on the Edit menu.**
- ➔ **If you delete a device connected between two devices, the connection between the two remaining devices is automatically preserved.**
A typical example would be if you have an effect device, connected as an insert effect between a synth and a mixer. If you delete the effect, the synth will be routed directly to the mixer.
- ➔ **When you move a device, connections are not affected.**
If you instead would like the program to re-route the device according to its new location in the rack, hold down [Shift] when you move it.
- ➔ **When you duplicate devices (by dragging) or use copy and paste, the devices are not auto-routed at all.**
If you would like them to be automatically routed, hold down [Shift] when you perform the operation.

Bypassing Auto-Routing

- If you wish to create a new device, without any auto-routing taking place, press [Shift] when creating the device.

Manual Routing

By selecting “Toggle Rack Front/Rear” from the Options menu or pressing [Tab] you turn the rack around. On the back of each device you will find connectors of two different types: audio and CV. As mentioned before, audio inputs and outputs are shown as large “quarter inch” jacks, while CV input and output jacks are smaller.

There are two ways to route audio from one device to another:

- By connecting “virtual patch cables” between inputs and outputs.
- By selecting connections from a pop-up menu.

Using Cables

! For the cables to be visible, the option “Show Cables” must be activated on the Options menu. See below.

1. Click on the desired input or output jack on one of the devices, and drag the pointer away from the jack (with the mouse button pressed).

A loose cable appears.



2. Drag the cable to the jack on the other device.

When you move the cable end over a jack of the correct type (audio/CV, input/output) it will be highlighted to show that a connection is possible.

3. Release the mouse button.

The cable is connected. If both input and output are in stereo and you connect the left channels, a cable for the right channel is automatically added.

- You can change an existing connection in the same way, by clicking on one end of the cable and dragging it to another connector.

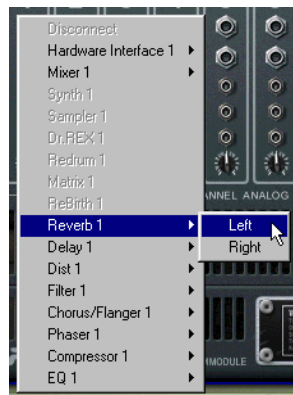
Using pop-up menus

1. Click (or right-click) on a connector.

A pop-up menu appears, listing all devices in the rack.

2. Move the pointer to the desired device (the device to which you want to create a connection).

A submenu appears, listing all suitable input/output connections. For example, if you clicked on an audio output on a device, the hierarchical submenus will list all audio inputs in all other devices.



- If a device is greyed out on the pop-up menu, there are no connections of the suitable kind.

3. Select the desired connector from the submenu.

The connection is created.

Disconnecting Devices

Again, there are two ways to disconnect devices:

- Click on one end of the cable, drag it away from the jack and drop it anywhere away from a jack.

or

- Click on one of the connectors and select “Disconnect” from the context menu that appears.

Using CV and Gate

CV/Gate is used for modulating and triggering device parameters. Each separate Device chapter lists the available CV/Gate connections, the parameters that can be modulated or be used for modulation outputs for the device.

Routing CV and Gate

There are not really any hard and fast “rules” applicable to CV/Gate routing. A few points should be mentioned though:

- **The specific “Sequencer Control” inputs present on the Subtractor, Malström, NN-19 and NN-XT sampler devices are primarily intended for controlling these devices as (monophonic) instruments from the Matrix Pattern Sequencer.**

If your intention is to use the Matrix CV/Gate outputs to create melodic patterns using these Instrument devices, you should use the Sequencer Control inputs.

- ★ **The Matrix Pattern Sequencer can be used in many other ways, besides creating melodic patterns. For example you could use it to modulate any CV controllable parameter, with the added advantage of the modulation being synchronized to the tempo.**

- **Conversely, if you would like to apply Gate or CV modulation to more than one voice, you should *not* use the Sequencer Control inputs, as these only function monophonically.**

- **Feel free to experiment: Use Gate signals to control parameter values and CV signals to trigger notes and envelopes, if you like.**
See the chapter “Matrix Pattern Sequencer” for more tips about using CV.

About the Voltage Trim Knobs

All CV inputs have an associated Trim knob. This is used to set the CV “sensitivity” for the associated parameter. The further clockwise a voltage trim knob is set, the more pronounced the modulation effect.

- Turned fully clockwise, the modulation range will be 100% of the parameters range (0-127 for most parameters).
- Turned fully anti-clockwise, no CV modulation will be applied.



REASON

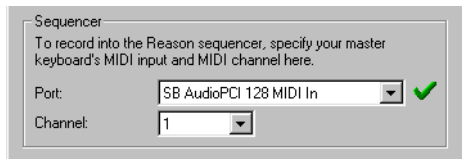
3

→ Routing MIDI to Reason

About the Various MIDI Inputs

All MIDI Inputs are set up in the Preferences-MIDI and Preferences-Advanced MIDI dialogs. This chapter describes the various ways you can set up how incoming MIDI is received.

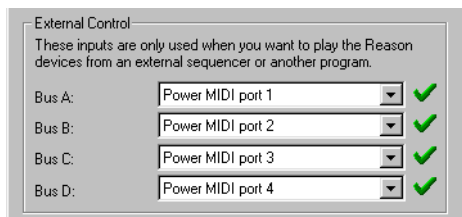
Sequencer Input



This is set in the Preferences-MIDI dialog. The Sequencer is the “standard” port for receiving MIDI input. This is what you should be using if you intend to use the Reason sequencer.

Once you have selected your MIDI interface on the Sequencer Port pop-up (and which channel it should receive on), you can direct incoming MIDI to any device by just clicking the “In” column to the left of a track name in the track list.

External Control Bus Inputs

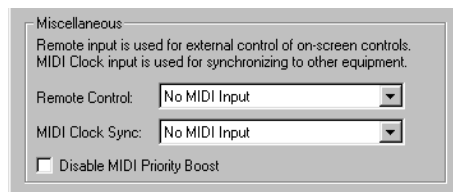


This is set in the Preferences-Advanced MIDI dialog. The External Bus inputs provide up to 64 MIDI input channels divided into four buses, each with 16 channels.

→ **These MIDI inputs are primarily for controlling Reason Devices from an external sequencer.**

This could be an external hardware sequencer or sequencer software that is installed on the same computer as Reason. You should preferably use a multiple port MIDI interface, so you can select separate ports for Reason and the other MIDI devices to use, although this isn't strictly required. See “Sending MIDI Data to Reason” below for further information.

Remote Control and MIDI Clock Input



These are set in the Preferences-Advanced MIDI dialog.

- The Remote Control input is used for assigning a MIDI port for receiving MIDI Controller messages for “live” remote control. How to use Remote Control is described in the chapter “MIDI and Keyboard Remote Control”.
- Using MIDI Clock, you can slave (synchronize) Reason to hardware devices (tape recorders, drum machines, stand alone sequencers, workstations etc.) and other computer programs running on the same or another computer. MIDI Clock is a very fast “metronome” that can be transmitted in a MIDI Cable. As part of the MIDI Clock concept there are also instructions for Start, Stop and locating to sixteenth note positions.

→ **By first selecting the appropriate MIDI input using the MIDI Clock pop-up and then selecting “MIDI Clock Sync” on the Options menu, Reason is ready to receive MIDI Clock sync.**
See the “Synchronization” chapter for more information.

Sending MIDI Data to Reason

Setting up MIDI Inputs under Mac OS 9

Under Mac OS 9, Reason requires OMS to receive MIDI. How to install OMS is described in the chapter "Installation" in the Getting Started book. OMS uses a concept of Devices, which basically means an external MIDI keyboard or sound module etc.

- **Each of Reasons' seven MIDI inputs can receive data from one OMS device.**

Use OMS Setup to create the devices needed (for example one per input).

- **One OMS device can be used for several of Reason's MIDI inputs.** However, note that this may lead to some confusion about what MIDI signals go where.

⚙ **We recommend that you use separate OMS Devices to each MIDI Input in Reason.**

- **If you have several MIDI programs running at the same time, they can share MIDI ports between them. Again, this might lead to confusing results and is probably best avoided.**

- ! **Please try to make sure that MIDI data sent to Reason is sent to Reason only, and not to any other application running at the same time.**

Setting up MIDI Inputs under Mac OS X

If you're using Mac OS X, there is no need for OMS. Reason instead makes use of the "CoreMIDI" services in Mac OS X, which eliminates the need for OMS.

- **For some MIDI interfaces connected via USB, no driver installation is required. Just plug in the interface and you're ready to go!**
- **For other, more advanced MIDI interfaces (or at least to take advantage of more advanced features, like multiple inputs) you will need to install a driver. Please consult the documentation that came with the interface for details.**

Setting up MIDI Inputs under Windows

In the Preferences-MIDI and Advanced MIDI dialogs, each MIDI input pop-up will show all MIDI input ports currently installed in your system.

Each of Reasons' seven MIDI inputs can receive data from any port. It is possible to route several MIDI inputs so that they receive data from the same physical MIDI In port, but you should generally avoid this as it can easily get confusing.

- **Reason only "grabs" the MIDI inputs you are actually using. MIDI inputs not selected in the Preferences – MIDI dialog are available to other programs.**

- **Note that other MIDI programs may "grab" all MIDI ports in your system when you launch them! If no MIDI inputs are available to Reason on startup a warning alert will appear.**

However, some of these programs allow you to disable the use of a particular MIDI input. If for example you have two MIDI interfaces, you may be able to set things up so that one of them is used by Reason and the other by the other application. Please consult the documentation for the other application for details.

Sending MIDI Data from Other Applications

Using ReWire 2

The preferred method for sending MIDI data into Reason from another application is by using ReWire (version 2 or later). In this way no additional system extension or utility is required, simply launch the applications and set them up so that MIDI is transmitted from the host (the “other” program) to the slave (Reason).

More about this on [page 47](#).

Mac OS 9 - Using OMS

If the application you want to use together with Reason is not ReWire compatible or if it only is compatible with ReWire version 1, you can instead use OMS to send MIDI from the application into Reason. To do this you need to have the OMS IAC (Inter Application Communication) driver installed.

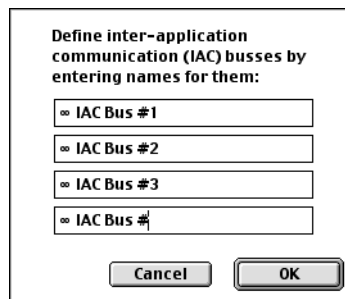
! Note that the IAC driver is *not* installed with the “Easy Install” option in the OMS installer. If you have installed OMS using this option, you need to perform a Custom installation, where the IAC driver can be selected (ticked) separately.

Installing more than one IAC Port

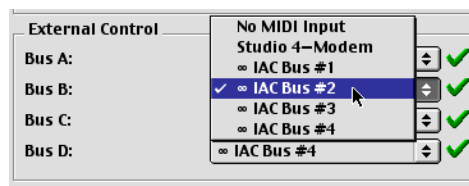
Once the IAC driver is installed, it is shown in your OMS Studio Setup dialog. Up to four IAC ports can be defined.

1. **Double click on the IAC Driver symbol.**
2. **Name as many ports as you require (up to four).**
3. **Close the dialog.**

4. Save your OMS Studio Setup.



The OMS IAC port naming dialog.



Selecting IAC busses in Reason's MIDI Preferences.

Setting up communication between two applications

Proceed as follows:

1. **Open the OMS MIDI Setup dialog in OMS Setup and make sure that “Run MIDI In Background” is enabled.**
2. **Launch Reason.**
It is important that you launch Reason after making changes to OMS, or the changes you have made will not be available.
3. **Set up the other program so that it transmits MIDI to an OMS IAC port.**
4. **In Reason, open the MIDI section of the Preferences dialog.**
5. **Open the MIDI input pop-up for the MIDI Input port(s) that should receive the incoming MIDI, and select the IAC port that you set up in step 3.**

Note that the Sequencer Port only receives MIDI on one selected channel at a time.

Mac OS X

As of this writing, the only way to route MIDI between applications in any practical way is to use ReWire 2. See [page 47](#).

Windows

If the application you want to use together with Reason is not ReWire compatible or if it only is compatible with ReWire version 1, you need to install some third part MIDI routing utility to be able to send MIDI from the application into Reason.

However, since such utilities are non-standard additions to the operating system, there is no guarantee that they will provide MIDI with reliable timing.

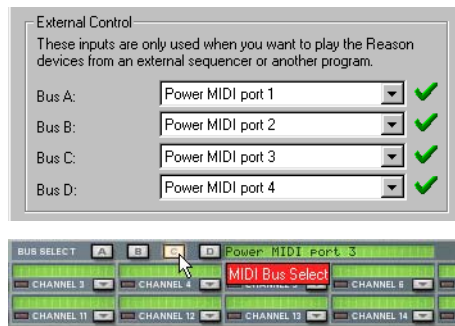
Please refer to the documentation that comes with the utility for detailed instructions.

Controlling Devices directly via MIDI

Routing MIDI to Devices

Depending on your MIDI interface, up to four separate ports, each with 16 channels, can be routed to Reason's External Control inputs. The following applies regarding setting up the External Control buses:

- **One port/device can be routed to each separate Bus input.**
Simply select the port/device using the appropriate Bus pop-up menu in the External Control section. One port/device can be routed to several Bus inputs.
- **When you have routed several MIDI ports/devices to corresponding External Buses, you use the Bus Select switch in the MIDI In Device to select a Bus (A-D) for editing the channel to device routing etc.**



If you would like to use an external sequencer to control Reason, there are basically two scenarios that could apply:

- **You have a “stand-alone” hardware sequencer or sequencer software installed on another computer.**

In this case, you should route the MIDI output from the sequencer (or the MIDI interface on the “other” computer) to the MIDI input on the interface connected to Reason. You should choose the External Control bus inputs for the incoming MIDI. This data is then routed to devices via Reason's MIDI In device.

→ **You have sequencer software installed on the same computer as Reason.**

This requires the "OMS IAC Driver" under Mac OS 9, or a MIDI routing application under Windows, as explained previously in this chapter.

→ **If you want to manually play (i.e not recorded MIDI data) Reason devices in real time from inside another sequencer program, MIDI thru must be activated.**

MIDI thru is when incoming MIDI is echoed out via the MIDI output. If you don't know how to do this, refer to the program's documentation. You will also need to make sure that the other application is "thruing" it's data to the correct MIDI port and on the right MIDI channel.

Bypassing the sequencer completely

It is possible to use Reason devices purely as "sound modules", bypassing the Reason sequencer completely. To do this, you should use the External Control busses to receive MIDI, and deselect the Sequencer port in the Preference dialog.

Once you have set up communication between Reason and the other device or application, you can hide the sequencer from view, by clicking the "maximize rack" button at the top of the rack's vertical scrollbar.

Sending Controller Data via MIDI

It is possible to send controller data from an external sequencer to control Reason parameters. Just set up your external device to transmit the correct MIDI controller messages on the right MIDI channel.

To find out which MIDI Controller number corresponds to which control on each device, please see the "MIDI Implementation Charts.pdf" document.

Once you have located the controller numbers and set everything up, you can record and edit the controller data in the external sequencer as you normally do, and the Reason parameters will react correspondingly.

⚙ **Do not confuse Remote Control and direct MIDI control. MIDI Remote allows you to map any MIDI Controller to any control on the front panel, but is primarily intended for "live" tweaking of parameters during playback.**

Recording Pattern changes

As specified in the MIDI Implementation, MIDI Controller #3 can be used to switch patterns in a device. However, pattern changes activated this way occur immediately (not at the end of the bar), which may or may not be what you prefer.

Please see [page 10](#) for information on recording and editing pattern changes.



REASON

4

→ Using Reason as a ReWire Slave

About this Chapter

This chapter describes how to use Reason as a ReWire slave, that is with Reason delivering audio to another ReWire compatible application. It does not deal with using ReBirth and Reason together; that is described on [page 208](#).

Why use Reason with ReWire?

While Reason is a complete music tool in its own right, you might want to add other elements to the music, such as:

- Vocals.
- Instrumental recordings.
- Hardware synthesizers (controlled via MIDI).

Connecting Reason to another application allows you to do just this, integrate your Reason songs with any other type of music, external MIDI and acoustic recordings. By recording Reason onto audio Tracks in an audio sequencer you can also continue processing your Reason tracks with other internal and external effects.

Introducing ReWire!

To make this integration between two audio programs possible, Propellerhead Software has developed ReWire. This technology provides the following possibilities and features:

In ReWire version 1

- Real time streaming of separate audio channels, at full bandwidth, into another audio program.
- Automatic, sample accurate, synchronization between the audio in the two programs.
- The possibility to have the two programs share one sound card.
- Linked transport controls that allows you to play, rewind etc, from either program.
- Less total system requirements than when using the programs together in the conventional way.

In ReWire 2

A number of features were added in Reason version 2. The following are the most important:

- Up to 256 audio channels (previously 64).
- Bi-directional MIDI communication of up to 4080 MIDI channels (255 devices with 16 channels each)
- Automatic querying and linking features that (among other things) allow a host to display the slave's devices, controllers, drum sounds etc. by name.

How Does it Work?

Basically the key to ReWire is the fact that Reason is divided into three components:

- The Reason application.
- The Reason Engine (a DLL on the PC and a Shared Library file on the Macintosh. Both located in the Reason program folder.)
- ReWire (also a DLL on the PC and a Shared Library on the Macintosh).

ReWire and the Reason Engine are common resources to the two programs (the other application and Reason) that generate the audio and passes it onto the other audio application.

! A note for Mac OS 9 users! In addition to being located in the Reason program folder, an alias for the Reason Engine is also put in the Extensions folder. This allows you, if required, to remove the alias. This will prevent Reason from running in ReWire mode, but it will still run fine as a freestanding application.

Terminology

In this text we refer to Reason as a ReWire *slave* and the application receiving audio from Reason (this could be Steinberg Cubase, Emagic Logic Audio or Mark of the Unicorn Digital Performer for example) as the *host* application.

About System Requirements

To run Reason together with another audio application of course raises the demands on computing power. However, adding ReWire to the equation does not in itself require a more powerful computer. On the contrary, it is likely that ReWiring two programs requires less power than for example running them with one audio card each.

Still, you should be aware that running two powerful audio applications on one computer will require a fast processor and most of all a healthy amount of RAM.

Preparations for Using ReWire - Mac OS 9 only

When you use ReWire, some of the system resources normally occupied by Reason are “transferred” to the other audio application: More specifically, the RAM required for loading samples in Reason, must now be provided by the host application instead. Therefore, when using ReWire we recommend you to make the following changes to your memory settings for the two programs (for details, see your Macintosh manual):

- 1. If you have raised the maximum memory setting in Reason (to be able to use more samples) lower it back to the recommended value, but make a mental note of how much it was set to.**
- 2. Raise the Maximum memory for the host application by at least the amount that you just lowered for Reason.**

Launching and Quitting

When using ReWire, the launch and quit order is very important:

Launching for normal use with ReWire

- 1. First launch the host application.**
- 2. Then launch Reason.**

Quitting a ReWire session

When you are finished, you also need to Quit the applications in a special order:

- 1. First quit Reason.**
- 2. Then quit the host application.**

Launching the host application for use without Reason/ ReWire

If you don't plan to run Reason, just launch the host application as usual. We recommend that you then also deactivate all ReWire channels if required (see the relevant section for your program, below). But this is not completely critical, ReWire does not use up very much processing power when it isn't used.

Launching Reason for use without the host application

If you want to use Reason as it is, without ReWire, just launch it as you normally do.

Launching both programs without using ReWire

We don't know exactly why you would want to run Reason and a ReWire host application at the same time on the same computer, without using ReWire, but you can:

- 1. First launch Reason.**
- 2. Then launch the host application.**

You may get a warning message in the host application, regarding ReWire, but you can safely ignore it. Please also note that the two programs now compete for system resources such as audio cards, just as when running either with other, non-ReWire, audio applications.

Using the Transport and Tempo Controls

Basic Transport Controls

When you run ReWire, the transports in the two programs are completely linked. It doesn't matter in which program you Play, Stop, Fast Forward or Rewind. Recording, however, is still completely separate in the two applications.

Loop Settings

The Loop in Reason and the corresponding feature (Loop, Cycle etc) in the host application are also linked. This means that you can move the start and end point for the Loop/Cycle or turn the Loop/Cycle on/off in either program, and this will be reflected in the other.

Tempo Settings

As far as tempo goes, the host application is always the Master. This means that both program will run in the tempo set in the host application.

However, if you are *not* using automated tempo changes in the host application, you can adjust the tempo on the transport in either program, and this will immediately be reflected in the other.

! If you are using the automated tempo changes in the host application, do not adjust the tempo on the Reason transport, since that tempo the doesn't have any effect on playback!

Synchronization

All synchronization to other equipment is handled from the host application, not Reason. In fact there are no special synchronization issues. All that is said in the host application's documentation about synchronizing audio channels is true for ReWire channels as well.

Routing Audio

Preparations in Reason

When you route audio from Reason to a ReWire host application, you make use of the Hardware Interface at the top of the rack. Basically, each output in the Hardware Interface is connected to a separate ReWire channel. Therefore:

→ **To take full advantage of the mixing features in the host application you need to connect the different Reason devices directly to the Hardware Interface.**

For example, if your Reason Song contains eight different instrument devices and you connect these to separate inputs on the Hardware Interface, they will appear on separate ReWire channels in the host application. You can then use the mixing facilities in the host application to adjust volume and pan, add effects and equalizing etc. - individually for each Reason device! If you instead connect all your Reason devices via a Mixer to the stereo input pair on the Hardware Interface, all sounds will appear mixed on a single ReWire stereo channel pair. While this works perfectly fine, you won't be able to mix and process the devices separately in the host application.

Routing in the ReWire host application

The following description is based on using Reason with Cubase SX as the host application. For descriptions on how to activate and route ReWire channels in other host applications, please go to www.propellerheads.se/rewirehelp.

1. **Pull down the Devices menu in Cubase SX and select the menu item with the name of the ReWire application (in this case Reason). All recognized ReWire compatible applications will be available on the Devices menu.**

The ReWire panel appears. This consists of a number of rows, one for each available ReWire channel.



2. **Click on the green buttons in the "Active" column to activate/deactivate the desired channels.**

The buttons light up to indicate activated channels. How many and which channels you need to activate depends on to which Hardware Interface inputs you have connected your Reason devices, as discussed above.

3. If desired, double click on the labels in the right column, and type in another name.

These labels will be used in the Cubase SX/SL Mixer to identify the ReWire channels.

4. Open the Cubase SX Mixer.

You will find that new channels have been added - one for each activated ReWire channel. If the channels aren't visible, you may need to scroll the Mixer window or check the Mixer View options (different channel types can be shown or hidden as desired in the Cubase SX Mixer).

5. Start playback (in Reason or Cubase SX - it doesn't matter as both programs will automatically be synchronized).

You will now see the level meters moving for the playing ReWire channel, and hear the sound of the Reason devices through Cubase SX's Mixer. Of course, this requires that your Reason Song contains some music!

6. Use the mixing features in Cubase SX to add effects, EQ, etc.

Routing MIDI via ReWire 2

The following description is based on using Reason with Cubase SX as the host application. For descriptions on how to route MIDI to Reason from other host applications, please go to www.propellerheads.se/rewirehelp.

1. In Cubase SX, select a MIDI track that you want to route to a device in Reason.

2. Pull down the MIDI Output menu for the track (in the Inspector or track list).

All devices in the current Reason Song are listed on the pop-up menu, along with the conventional, "physical" MIDI outputs.

3. Select a Reason device from the pop-up menu.

The output of the MIDI track is now routed to that device.

- If you now play back a MIDI part on the track, the MIDI notes will be sent to the Reason device - just as if the track were connected to any regular MIDI sound source.

The sound of the device will be sent back into Cubase SX via ReWire - which channel it will appear on depends on how you have routed the device to the Hardware Interface in Reason, as discussed above.

- To play the device "live", you need to select the proper MIDI input for the track in Cubase SX (the input to which your MIDI keyboard is connected) and activate the Monitor button for the track.

When the Monitor button is activated, all incoming MIDI (i.e. what you play on the keyboard) is immediately sent to the track's MIDI Output (i.e. to the Reason device).

Converting ReWire Channels to Audio Tracks

Most often, there is no need to convert individual ReWire channels to regular audio tracks! The channels already appear in the host application's Mixer, and you can typically perform the same kind of real-time processing as with regular audio channels (effects, EQ, volume, pan and mute automation, etc.).

Still, you may need to convert the ReWire channels to audio tracks, for example if you want to continue working in Cubase SX only. This is probably easiest done by using the host application's "Export Audio" or "Bounce" function. In Cubase SX, you would proceed as follows:

1. Make sure your Reason devices play back properly via ReWire.

2. In the Cubase SX Mixer, activate Solo for the ReWire channel you want to convert to a regular audio track.

Make sure no other channel is Soloed as well.

3. Go to Cubase SX's Project window and set the left and right locator to encompass the whole song (or a section, if that's what you want). Make sure the Cycle (loop) function is turned off.

4. Pull down the File menu in Cubase SX and select "Audio Mixdown" from the Export submenu.

The Export Audio Mixdown dialog appears.

5. Activate the "Import to Pool" and "Import to Track" options and fill in the rest of the dialog as desired.

You can choose to include any Cubase SX mixer automation, select a file format and file name, etc.

6. Click Save.

The ReWire channel is now rendered to a new audio file on disk. A clip referring to the file will appear in the Pool, and an audio event playing this clip will be created and placed on a new audio track, starting at the left locator.

- If you now play back the audio track you will hear exactly what was played on the ReWire channel.

This means you should keep that ReWire channel muted (or deactivated) now, since otherwise you would hear the sound twice - once via ReWire and once from the audio track.

- To convert all your ReWire channels this way, simply proceed as above (but solo another ReWire channel in the Cubase SX Mixer).

- ! Converting ReWire channels this way results in a number of audio files that can be very large (depending on the length of the song). Make sure you have enough disk space!

Details About Various ReWire Hosts

The Propellerhead Software website provides updated information on how to configure ReWire for most compatible host applications. Please go to:

www.propellerheads.se/rewirehelp.



REASON

5

→ MIDI and Keyboard Remote Control

Introduction

It is possible to assign computer keyboard commands and/or MIDI controller messages to most Reason device parameters or functions. Both methods allow you to use a “learn” function to instantly assign the parameter knob, slider or button to a keyboard command or a controller on an external device.

MIDI Remote Mapping

If you want to control one or several Reason parameters in real time from an external MIDI device, you can use MIDI Remote Mapping. The external device could be a dedicated MIDI performance controller, for example.

Setting Up

! If you are using a single MIDI interface with only one MIDI Input, there are certain restrictions to how you can use MIDI Remote Mapping. See “Example MIDI Setups” below for an explanation.

To set up Reason for MIDI Remote Mapping, proceed as follows:

- 1. Open the Preferences dialog from the Edit menu and select the Advanced MIDI page.**
- 2. Open the Remote Control pop-up in the Miscellaneous section, and select your MIDI input.**

This should preferably be a separate port that you “dedicate” to sending controller messages, see below.

- 3. Close the Preferences dialog.**

Example MIDI Setups

There are several possible variables when it comes to what type of MIDI setup you are using. Please read on.

“Ideal” Setup

The ideal setup is a computer with two MIDI interfaces or one MIDI interface with multiple, separate inputs, a MIDI keyboard used for playing/recording and a separate MIDI Controller device (“fader box”) used for remote control.

1. **Connect your MIDI keyboard to one MIDI Input.**
2. **Connect your MIDI remote device to the other MIDI Input.**
3. **Open the Preferences – MIDI dialog and set things up so that the MIDI keyboard is used for playing and the fader box is used for MIDI Remote control.**

If You are Using a Single MIDI Interface with one MIDI Input only

In this scenario we assume you have your MIDI keyboard and external MIDI controller connected to the *same* MIDI Input on your computer. In this case it's a little bit trickier to get things to work correctly. Here's the problem:

If you are using the sequencer input for playing a device, the device will react to MIDI controller messages via this input, since all devices are always set up to react to MIDI controller messages (see [page 287](#) for details).

Now, if you happen to set things up so that a control on a device reacts to the *same* MIDI Controller message as is used for remote control of *another* control (maybe even on another device) both controls will move simultaneously on screen!

The solution is to separate things via MIDI Channel messages. Proceed as follows:

1. **If you are using the MIDI sequencer input to play your devices, open the Preferences - MIDI dialog and make a note of which MIDI Channel is used for sequencer input.**
2. **If you are instead using one of the direct MIDI buses A to D, check the Hardware Interface to find out which MIDI Channels are already taken by devices in the rack.**
3. **Set up your MIDI Controller (that you plan to use for MIDI Remote control) to transmit on any MIDI channel that is not already occupied (as described above).**
4. **When you then set up MIDI remote Control, only use the MIDI Channel on which your MIDI Controller is now set to transmit on.**
This will ensure that remote control doesn't conflict with other MIDI.

If you only have one MIDI Device

If you only have one MIDI Device that you plan to use both for playing/recording and for remote control, there are severe restrictions. Actually there 's only one sensible way to avoid conflicts.

1. **Open the “MIDI Implementation Charts.pdf” document.**
This can be found in your program folder.
 2. **Make a note of the controller numbers that are not used for direct control of *any* device at all.**
 3. **Set up your MIDI remote control so that it only uses these unused MIDI Controller numbers.**
- ! **Please note that you can only assign a MIDI Controller number for remote control of one parameter at a time. If you try to assign a second parameter to a MIDI Controller number already used, the previously assigned parameter is overridden by the new one.**

Enabling MIDI Remote

To enable MIDI Remote, select “Enable MIDI Remote Mapping” from the Options menu.

Editing MIDI Remote Mapping

1. To get an overview of which parameters are MIDI remote controllable select **"Edit MIDI Remote Mapping"** from the Options menu.

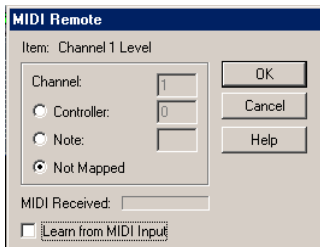
When done, each device you select will show a green arrow symbol beside every parameter that can be assigned a MIDI remote.



A section of a mixer with MIDI Remote enabled.

2. If you click on a assignable parameter, a dialog appears allowing you to select a MIDI controller (or a Note number) to control that parameter.

Note numbers function exactly like Keyboard remote - they can only control on/off or min/max values (see [page 57](#)).



The MIDI Remote dialog.

3. Make sure that the **"Learn from MIDI Input"** box is ticked.

4. Simply turn the knob (or slider etc.) that you wish to use to remote control the parameter.

The "MIDI Received" field momentarily flickers as you turn the knob, and then the dialog shows the controller number and the channel it is transmitted on.

5. Click **"OK"** to exit the dialog.

The selected parameter now has a tag, displaying the controller number, and the MIDI channel used.

6. To exit Edit MIDI Remote Mapping mode, deselect it from the Options menu.

You do not always have to use this method - see below.

About the two Edit MIDI Remote Mapping Modes

If Edit MIDI Remote Mapping is enabled (ticked) on the Options menu, assigned parameters are "tagged", and the arrow indicators show the assignable parameters. In this mode, however, you cannot operate Reason normally, as every parameter you click on will open the MIDI Remote dialog. The Edit mode is primarily for overview of available parameters and the current assignments.

- Another way to assign keyboard remote commands is to have **"Edit MIDI Remote Mapping" deselected on the Options menu, and to simply [Ctrl]-click (Mac) / right-click (PC) the parameter you wish to remote control.**

This opens a pop-up menu, where one of the options will be "Edit MIDI Remote Mapping". Selecting this opens the MIDI Remote dialog. Thus, you do not have to select Edit mode from the Options menu if you already know that a parameter is free and assignable.

Keyboard Remote

Assigning keyboard remote commands is very similar to MIDI remote mapping. However, as there is no MIDI involved, there is no special setting up required. Keyboard commands can be assigned the same parameters as when using MIDI remote mapping, but the functionality differs in one central aspect:

- **Keyboard Remote commands can only be used to toggle on/off or min/max values for an assigned parameter.**

Hence, if you assign a keyboard remote command for a knob, slider or spin control, it will only switch between the minimum and maximum values for that parameter. The only exception to this are the multi-selector buttons used for various parameters such as envelope destination, for example. These will cycle through the available options when using keyboard remote.

Enabling Keyboard Remote

To enable Keyboard Remote, select “Enable Keyboard Remote” from the Options menu, or press [Command]+G (Mac) or [Ctrl]+G (PC).

Editing Keyboard Remote

- **To get an overview of which parameters are remote controllable select “Edit Keyboard Remote” from the Options menu.**

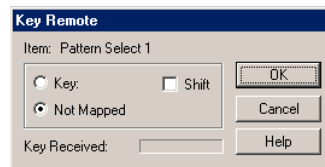
When done, each device you select will show a yellow arrow symbol beside every parameter that can be assigned a keyboard remote.



A section of a drum machine with Keyboard Remote enabled.

- **If you click on an assignable parameter, a dialog appears allowing you to select a key command for that parameter.**

You may use any key except the Space bar, Tab, Enter or the Numeric keypad (which is reserved for Transport functions) or a combination of [Shift] + any key (with the same aforementioned exceptions).



The Keyboard Remote dialog.

- **Simply press the key (or key combination) you wish to use to remote control the parameter.**

The “Key Received” field momentarily indicates that it is “learning” the key-stroke(s), and then the dialog displays the name of the key you have pressed. If [Shift] was used, the box beside the word Shift in the dialog is ticked.

About the two Edit Keyboard Remote Modes

If Edit Keyboard Remote is enabled (ticked) on the Options menu, assigned parameters are “tagged”, showing the remote key for that parameter. In this mode, however, you cannot operate Reason normally, as every parameter you click on will open the Key Remote dialog. This mode is primarily for overview of available parameters and the current assignments.

- **Another way to assign keyboard remote commands is to have “Edit Keyboard Remote” deselected on the Options menu, and to simply [Ctrl]-click (Mac)/right-click (PC) the parameter you wish to remote control.**

This opens a pop-up menu, where one of the options will be “Edit Keyboard Remote”. Selecting this opens the Key Remote dialog. Thus, you do not have to enable/disable Edit mode from the Options menu if you know that a parameter is assignable.

- ! **If you try to assign a Remote Key that is already in use, you will get an alert asking if you wish to change the current assignment.**

Saving Remote Setups

MIDI or Keyboard Remote setups are always stored with the song. But perhaps you wish to be able to recall this setup for use in a new song, or permanently use a specific remote setup.

- **This could be done by saving a song document containing all the devices that are affected by the remote setup together with the related Key or MIDI mapping, but without any sequencer data.**
This song document could then be used as a starting point for any new song, by simply loading it, and immediately using “Save As” to save it under a new name.



REASON

6

→ Synchronization

ReWire users – Read This!

This chapter is about synchronization via MIDI Clock, and does not apply to users of ReWire. If you are using Reason together with a ReWire compatible application, ReWire automatically handles all synchronization issues for you. See [page 47](#) for details.

What is Synchronization and MIDI Clock?

Synchronization, in this context, is when you make Reason play at the same tempo as another device; where both start, stop and can locate to certain positions, together. This is done by transmitting MIDI Clock signals between Reason and the other device. MIDI Clock is a very fast “metronome” that can be transmitted in a MIDI cable. As part of the MIDI Clock concept there are also instructions for Start, Stop and locating to sixteenth note positions.

You can set up synchronization between Reason and hardware devices (tape recorders, drum machines, stand alone sequencers, workstations etc.) and other computer programs running on the same or another computer.

Master/Slave

In a synchronized system there is always one master and one or more slaves. In our case, the master is the one that controls the tempo. In other words, it is only the tempo setting on the master device that is of any relevance, since the slaves slavishly follow the master's tempo.

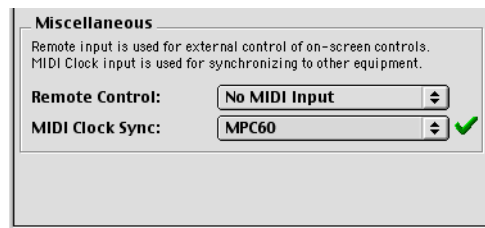
- **Reason always acts as a slave. That is, it receives MIDI Clock signals, it never transmits them.**
- ! **Before you create any serious projects that require sync, try out the features described below and check out “Synchronization Considerations” on page 117.**

Slaving Reason to an External Device

This example assumes that you have an external device, such as a drum machine, hardware sequencer, another computer, tape recorder etc., that transmits MIDI Clock signals to which you want to synchronize Reason.

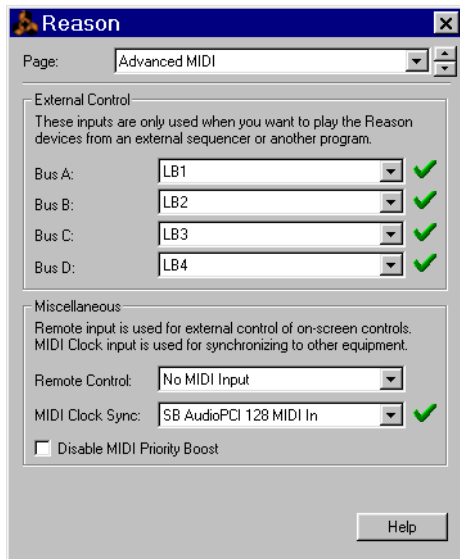
1. **Connect a MIDI Cable from the MIDI Out on the other device to a MIDI In on the computer running Reason.**
2. **Set up the other device so that it transmits MIDI Clock signals to the MIDI Out you just connected to the computer running Reason.**
3. **In Reason, pull down the Edit menu (under Mac OS X, the Reason menu) and open the Preferences dialog. Select the Advanced MIDI page.**
4. **Pull down the MIDI Clock Sync pop-up and select the MIDI Input to which you connected the MIDI cable from the other device.**

Under Mac OS 9, if you do not understand which Input this is, or if that MIDI input doesn't appear in the list, consult your OMS documentation for information on MIDI interfaces, MIDI ports and naming.



Reason Mac OS 9 set up to sync to MIDI Clock coming in from a MPC-60 drum machine connected to an external MIDI interface.

Under Windows, if you can't find the MIDI Input you want to use, there is either something wrong with the installation of the interface, or some other program is holding on to it. Consult the documentation for the MIDI interface, the other program and Windows, for more information.



Reason Windows set up to sync to MIDI Clock coming in via the MIDI interface on a SB PCI 128 card.

5. Close the dialog.

6. Activate MIDI Clock Sync from the Options menu in Reason.

This can also be set on the Transport panel.

7. Activate playback on the other device.

Reason will start playing 'in sync' with it and the Sync LED on the Transport will light up.

Slaving Reason to Another Program on the Same Computer

! The preferred method for synchronizing two applications is by using ReWire, see page 47. However, if the application you need to sync Reason with doesn't support ReWire, you can try the procedures described below.

This section describes how to use MIDI Clock to synchronize Reason to another application running on the same computer. This text is based on the following assumptions:

- Under Mac OS 9, that the other program has full support for OMS and that you have read and understood the instructions on MIDI via OMS in general, described in the chapter "Routing MIDI to Reason".
- Under Windows, that you have access to a MIDI routing utility, as described on page 45.

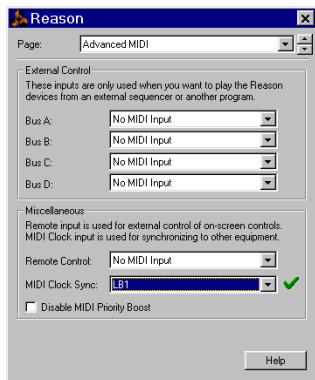
! Note that synchronizing via MIDI Clock makes the two programs play at the same time, that is, they both "run" when you "hit play". It does not mean they can both play audio at the same time. See page 281 for details about "sharing audio".

! A note for users of Mac OS X: As of this writing, there was no practical way of synchronizing two applications without ReWire.

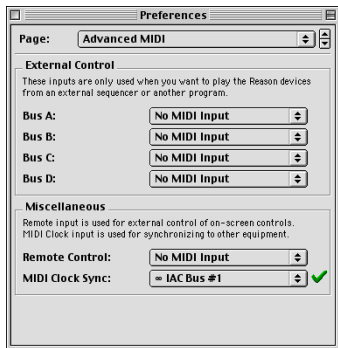
Proceed as follows:

- 1. Set up the other program, so that it transmits MIDI Clock to Reason:**
 - Under Mac OS 9 this is done by selecting the OMS IAC port.
 - Under Windows this is done by selecting one of the MIDI routing utility ports.
- 2. In Reason, pull down the Edit menu and open the Preferences dialog. Select the Advanced MIDI page.**

3. Pull down the MIDI Clock pop-up and select the corresponding MIDI routing utility port.



Windows: Reason set up to sync to MIDI Clock coming in via "Power MIDI" port 1.



Mac OS 9: Reason set up to sync to MIDI Clock coming in via OMS IAC.

4. Close the dialog.
5. Activate MIDI Clock Sync from the Options menu in Reason.
6. Activate playback on the other device.
Reason will start playing 'in sync' with it and the Sync LED on the Transport will light up.

Synchronization Considerations

Adjusting for Latency

Output Latency:	11 ms
Latency Compensation:	11 ms

Latency compensation.

Because of the latency problem described on page 282, you might need to adjust Reason's playback in relation to the sync master, so that they are in perfect time. The tempo will not differ between the two, but Reason might play ahead or behind the other application. You might need to adjust this. However, this is something you only need to do once. The setting is stored with your other preferences, so you don't need to adjust it again.

Proceed as follows:

1. Set up the other application so that it generates a solid click, on for example quarter or eighth notes, preferably with a special sound on the downbeat.
This click can either come from an internal metronome or from a MIDI source. If you use a MIDI source, make sure you pick one that has solid MIDI timing.
2. Set up Reason so that it plays a similar rhythm as the other application.
You might for example use the Metronome or Redrum drum computer for this.
3. Start the two applications in sync.
4. Make sure you hear both applications at approximately equal level.
5. Open the Preferences dialog in Reason and select the Audio page.
6. Trim the "Latency compensation" setting until the "clicks" from the both sources sound at exactly the same time.
7. Close the Preferences dialog in Reason.

If Latency Compensation isn't enough

There might be situations where you can't compensate enough in Reason to make two software applications run in sync. This might especially be true if the other application is an audio sequencer, that is if it can record and playback both audio and MIDI.

This problem is an indication of the fact that the other application has not been set up properly and that its audio playback is not in sync with its own MIDI playback.

! This is not something that you can or should compensate for in Reason. Instead, follow the instruction included with the other application to make sure its MIDI playback and audio playback are correctly locked to each other.

About the beginning of the Song

Due to the latency phenomenon, described on page 282, Reason needs some time to correct its playback speed when it first receives the Start command. This can be noted as a small glitch in the audio playback, when the program starts. If this is a problem, you need to insert a couple of empty measures at the beginning of the Song. Proceed as follows:

1. **Set the Left Locator to "1 1 1" and the right Locator to "3 1 1".**
2. **Click somewhere in the main sequencer area to move the menu focus to the sequencer.**
3. **Select "Insert Bars Between Locators" from the Edit menu.**
4. **Set up the other device/application, so that it also plays two empty bars at the beginning.**

About MIDI Song Position Pointers

MIDI Clock actually consists of five type of messages: The actual clock (the metronome that establishes the tempo), Start, Stop and Continue commands and Song Position Pointers. This last type of message contains information about positions, so that a program for example "knows" where in a Song to start playback from.

Normally, this ensures that you can locate to any position and activate playback from there. In older devices, Song Position Pointers might not be implemented. This means that you will be able to synchronize properly only if you start both devices from the absolute beginning of the song.

About Tempo Changes

Again, due to the latency phenomenon, Reason needs a bit of time to adjust to changes in tempo. If there are abrupt changes in the MIDI Clock, due to drastic tempo changes in the master, you will note that Reason will require up to one measure to adjust itself to the change. How long this actually takes also depends on the precision of the incoming MIDI Clock. The more precise it is, the faster Reason can adjust to it.

If this adjustment is a problem, try to use gradual tempo changes rather than immediate ones.

! When Reason is synchronized to MIDI Clock, there is no Tempo readout.



REASON

7

→ Optimizing Performance

Introduction

Reason is a program of infinite possibilities. You can create as complex songs as you like, using endless racks of devices. While this is one of the most exciting properties of the program it does have a drawback – it means that you must be careful with how you manage your computer processing power.

Each device you add to the rack uses up a bit of computer processing power – the more devices the faster the computer has to be. However, you can set up your devices to require more or less processing power. For example a sound on the Subtractor synthesizer that only uses one oscillator and one filter requires much less processing power than one using both dual oscillators and dual filters.

Samples used in your songs also require RAM - memory - to load properly. The use of RAM can also be managed, as described at the end of this chapter.

When creating songs for other people, for example for publishing in the Reason song archive (see www.propellerheads.se for more information), you should do what you can to reduce the requirements for playing back a certain song, both in terms of processing power and in terms of RAM requirements. Other users may not have as powerful a computer as you do!

Checking Processing Power

On the transport you will find a meter labelled CPU. This indicates how much processing power is used at any given moment.



The CPU meter.

The higher this meter goes, the higher the strain on your computer processor. You will note when your processor is heavily loaded that graphics will update slower. Finally, when there's too little power left to create the audio properly, the sound will start breaking up.

Optimization and Output Latency

As described on page 282, you generally want the lowest possible latency, to get the best response when you play Reason in real time. However, selecting too low a latency is likely to result in playback problems (clicks, pops, dropouts, etc.). There are several technical reasons for this, the main one being that with smaller buffers (lower latency), the average strain on the CPU will be higher. This also means that the more CPU-intensive your Reason song (i.e. the more devices you use), the higher the minimum latency required for avoiding playback difficulties.

Therefore, you may need to adjust the latency. This is done differently depending on which audio cards, drivers and operating system you are using:

Making adjustments in the ASIO Control Panel

If you are using an ASIO driver specifically written for the audio hardware, you can in most cases make settings for the hardware in its ASIO Control Panel. This panel (opened by clicking the ASIO Control Panel button in the Preferences-Audio dialog) may or may not contain parameters for adjusting the latency. Usually this is done by changing the number and/or size of the audio buffers - the fewer and smaller the audio buffers, the lower the latency. Please consult the documentation of your audio hardware and its ASIO drivers for details!

! Raising the buffer size to eliminate audio artefacts on playback is mainly effective if you are currently using very small buffers, 64 to 256 samples. If the buffers are already big (1024 or 2048 samples) you will not notice much difference.

Making adjustments in the Reason Preferences dialog

If you are running Reason under Windows and using an MME or DirectX driver, or if you are running Reason under Mac OS X and using a CoreAudio driver, you can adjust the output latency in the Preferences – Audio dialog.

- Under Windows and Mac OS X, this is done by dragging the Buffer Size slider.
- If you are running Reason under Mac OS 9.x using the Sound Manager Default Output driver, you cannot change the latency.

General procedure

The basic procedure for optimizing the latency is the following:

1. Open a song and start playback.

You want to choose a song that is reasonably demanding, i.e. with more than just a few tracks and devices.

2. Open the Preferences dialog.

Under Mac OS X, this is found on the Reason menu; under all other operating systems it's found on the Edit menu.

3. Select the Audio page and locate the buffer settings.

If you are using an ASIO driver, you need to click the ASIO Control Panel button, for Mac OS X/CoreAudio, Windows/MME or DirectX you should use the Buffer Size slider.

! If you are making adjustments in the ASIO Control Panel for hardware with an ASIO driver, you should make a note of the current buffer settings before changing them.

4. While the song is playing, listen closely for pops and clicks and try lowering the latency (buffer size/number).

5. When you get pops and clicks, raise the latency value a bit.

6. Close the Preferences dialog (and ASIO Control Panel, if open).

About Latency Compensation

In the lower right corner of the Preferences-Audio dialog, you will find a setting called Latency Compensation. This value is used internally in Reason to compensate for the latency when synchronizing Reason to another MIDI sequencer or similar. Usually, Latency Compensation is set to the same value as the Output Latency, but it is possible to increase it (see page 62). Normally however, you shouldn't need to touch this parameter.

Optimizing Your Computer System

In this manual we do not have the possibility to give you detailed procedures for optimizing your computer for maximum power. This is a subject that we could write complete books on! However, we'd like to share a couple of important tips:

→ **Quit other programs that are running at the same time as Reason.**

→ **Remove background tasks on your computer.**

This might be any background utilities you have installed as well as networking, background internet activities etc.

→ **Under Windows, make sure you use the latest and most efficient driver for your audio card.**

Generally, ASIO drivers are the most efficient, followed by DirectX and last MME.

→ **Only work on one Reason document at a time.**

Songs that are open in the background do consume some processing power even though they're not playing.

→ **Lower the sample rate setting on the Preferences dialog.**

While this also reduces sound quality, it is a very quick and convenient way to try to play a song that your computer otherwise can't handle.

→ **Make sure your computer display is set to 16-bit colors.**

Under Windows, this mode is called "High Color"; under Mac OS it is called "Thousands of colors".

Optimizing Songs

Below follows things you can check and change to make sure your song uses as little computer processing power as possible:

Global

- **Delete unused devices.**
If a device isn't actually doing anything, delete it from the rack.
- **Use fewer devices.**
For example, instead of using several reverbs as insert effects, replace them all with one, set up as a send effect. By the same token, try to use one sampler playing several different samples instead of numerous samplers playing one sample each.
- **Don't use stereo unless it is required.**
For example, if a sampler or Dr. Rex player is playing mono material, only connect the Left output and leave the Right output unconnected.

Sample Players – NN19, NNXT, Dr. Rex and Redrum

- **Only activate High Quality Interpolation when it is required.**
Listen to the sound in a context and determine whether you think this setting makes any difference. However, note that on a Macintosh G4, High Quality Interpolation does not require any more processing power.
- **If you are playing back a sample at a much higher pitch than it was recorded at, consider sample rate converting the actual sample file to a lower sample rate.**
This will require an external sample editor with good sample rate conversion facilities.
- **Try to refrain from using stereo samples.**

Filters – Subtractor, Malström, NN19, NNXT and Dr. Rex

- **Deactivate filters that are not used.**
Observe that if the Cutoff is all the way up or the envelope is set to open the filter fully, then the filter doesn't affect the sound. Conserve processing power by disabling the filter altogether.
- **Where applicable, use the 12dB lowpass filter instead of the 24dB lowpass filter.**
See if you can get the same sonic result by using the 12dB filter, since it uses up less processing power.

Polyphonic Devices – Subtractor, Malström, NN19, NNXT, Dr. Rex and Redrum

- **Try making the device play fewer voices.**
This can be done for example by lowering the release and setting the Polyphony setting to exactly the maximum number of notes played simultaneously by this device.
- ★ **Please note that just lowering the polyphony setting has no effect. Unused voices do not consume processing power.**
- **Where applicable, try the Low Bandwidth (Low BW) setting.**
This will remove some high frequency content from the sound of this particular device, but often this is not noticeable (this is especially true for bass sounds).

Subtractor

- **Try avoiding using Oscillator 2 altogether.**
If you can create the sound you need with only one oscillator, this saves considerable amounts of processing power.
- **Do not use the oscillator Phase mode if you don't need it.**
In other words, set the Oscillator Mode switches to "o", not "*" or "-".
- **Do not activate Noise unless required.**
- **Do not activate Filter 2 unless required.**
- **Do not use FM unless required.**
In other words, set the oscillator FM knob to "0" and make sure no modulation source is routed to FM.

Malström

- **If it isn't necessary, refrain from using Osc B at all.**
If you can produce the desired sound by using Osc A only, this will save a lot of processing power.
- **If one or both Oscillators are routed to one Filter only, and/or the Spread parameter is set to "0", only connect one of the outputs (the one to which the filter is connected) to the mixer, and leave the other one unconnected.**
- **Try to see if you can achieve the desired effect by using only one of the filters, and without using the shaper.**
Using both of the filters and the shaper in conjunction requires considerably more processing power than using just one of the filters and/or the shaper.

Redrum

- **Do not use the Tone feature available on channel 1, 2 and 9.**

In other words, make sure the Tone controls and their accompanying Vel knobs are set to "0" ("twelve o' clock").

Mixer

- **Avoid using stereo inputs when not required.**

For example, if your sampler or Dr. Rex player is playing mono material, only connect it to the Left input on a mixer channel. Leave the Right input unconnected.

- **Do not activate EQ unless required.**

If a channel doesn't make use of EQ, make sure it's EQ button is deactivated.

Distortion

- **The D-11 Foldback Distortion will use up less CPU power than the Scream 4 Distortion device.**

Reverb

- **The RV-7 uses much less power than the RV7000.**

For some applications the RV-7 might do just fine, and will use up much less power.

- **If you are running out of processing power, try the Low Density algorithm for the RV-7.**

This uses up much less power than other algorithms.

Send Effects

- **When you are using mono effects as send effects, you can connect the effect returns in mono as well (disconnect the cable to Aux Return Right on the Mixer).**

This is true for the following effects:

- D-11 Distortion.
- ECF-42 Envelope Controlled Filter.
- COMP-01 Compressor.
- PEQ-2 Parametric EQ.
- DDL-1 Delay (provided the Pan parameter is set to center position).

Songs and Memory Requirements

Songs not only use up system resources in terms of processing power, they also require RAM (memory) to load at all.

The amount of RAM required for loading a song, is directly proportional to the amount of samples used in the song. For example, a song only using Subtractors and effects requires very little RAM.

If you are running out of RAM try the following:

- **Close other song documents.**

All open songs compete for RAM

- **Under Mac OS 9; raise the memory setting for Reason.**

This is done in the Finder by selecting the Reason application and opening the Get Info window.

- **Under Windows or Mac OS X, terminate other applications.**

All running applications compete for the RAM available in the computer.

- **Use mono samples instead of stereo.**

Mono samples require half the amount of RAM.

- **Try sample rate converting sample files to a lower sample rate.**

Note that this will affect sound quality negatively. Also note that it will require an external sample editor with good sample rate conversion facilities.



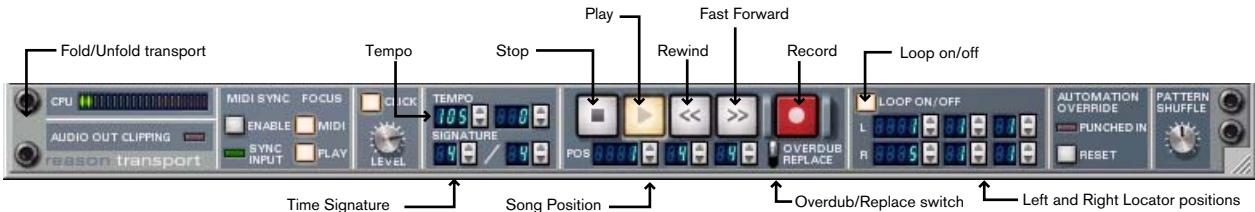
REASON

8

→ Transport Panel

Overview

The transport panel has standard controls for the sequencer transport, but also features controls for setting tempo, metronome click, locator points etc. The main controls in the central area of the transport panel are as follows:



Main Transport Controls

The main transport controls function just like standard controls on tape recorders etc. There are also fixed computer keyboard combinations for the most important transport functions:

Function	Key command	Comments
Stop	[0] on the numeric keypad or [Return]	Pressing Stop during playback stops the sequencer. Pressing stop again, sets the position to the left locator (if this is located before the current position). Pressing stop a third time sets the position to the start of Bar 1. The Stop button also sends out a "Reset" message, in case of stuck notes or other related problems.
Play	[Enter] on the numeric keypad	Starts playback of the sequencer.
Rewind	[7] on the numeric keypad	Clicking once moves the position backward one Bar. If you press and hold this button on the transport (not using key command) it will start scrolling faster after about 2 seconds.
Fast Forward	[8] on the numeric keypad	Clicking once moves the position forward one Bar. If you press and hold this button on the transport (not using key command) it will start scrolling faster after about 2 seconds.
Record	[*] on the numeric keypad, or [Command]/[Ctrl]-[Return]	Activates "Record ready" mode if sequencer is stopped. If activated during playback it will start recording immediately ("punch in").

You can also use the following transport related key commands:

Function	Key command	Comments
Toggle Stop/Play	Space bar	Switches between stop and play mode.
Go to Left Locator (Loop Start)	[1] on the numeric keypad	Sets the position to the left locator.
Go to Right Locator (Loop End)	[2] on the numeric keypad	Sets the position to the right locator.

Tempo and Time Signature



The tempo and time signature settings can be adjusted on the transport panel. The left tempo field sets the tempo in bpm, and the tempo field to the right allows you to fine tune the tempo, in steps of 1/1000 bpm.

- **You can specify any tempo between 1 and 999.999 bpm (beats per minute).**
- **You can also adjust the tempo (in bpm steps) by using the [+] and [-] keys on the numeric keypad.**
- **You set the time signature by specifying a numerator (left value field) and a denominator (right value field).**

The numerator is the number of beats per bar, and the denominator governs the length of a beat.

Song Position



The song position in Bars, Beats and 16th notes is shown (in that order) in the three fields below the transport controls. You can set the positions by using the spin controls.

- **You can also set the position by double clicking on a Pos value box, typing in a new position (in the format “Bars.Beats.16th notes”) and pressing [Return].**
If you only type one or two numbers, the remaining numbers will be set to their lowest values (e.g. type “5” to set the position to “5.1.1”)

Left and Right Locator Positions



The left and right locators are used for several things, like setting the boundaries of a loop or inserting/removing bars. You can set the positions for both locators by using the spin controls on the transport panel or by double clicking and typing a position value.

Loop On/Off

In loop mode, the sequencer will repeat a section over and over again, during playback or recording. You specify the section to be looped by setting the left and right locator.

Overdub/Replace Switch



When recording over a previously recorded section, this switch governs the following:

- **In Overdub mode, the new recording is added to whatever was on the Track before.**
- **In Replace mode, the new recording replaces any previously recorded notes.**
- ! **Note that controllers and pattern changes aren't affected by this - recording controllers will always replace any previously recorded controller values. However, you may still want to set the switch to Overdub mode, to avoid accidentally removing some recorded notes in the area.**

Additional Transport Panel Items

Click



When this is activated, you will hear a click on each beat, with an accent on the downbeat of each bar. The click is played back during recording and playback. You can adjust the volume of the click by using the Level knob.

MIDI Sync and Focus



This section of the Transport Panel contains items relating to MIDI sync.

- ➔ **The “Enable” button puts Reason into MIDI sync mode.**
The transport controls will be disabled, and Reason will not run unless MIDI sync data is provided from an external device.
- The MIDI and Play Focus buttons relate to how incoming MIDI and MIDI sync should be handled if there are several open Song documents. If you have two or more Songs opened, and no MIDI sync is used, the currently selected Song (the document “on top”) always has MIDI focus. If MIDI Sync is enabled (which is global for all currently open Song documents), this functionality changes in the following way:
 - ➔ **If both “Play” and “MIDI” are activated for a Song, incoming MIDI data and MIDI sync will be sent to this Song, regardless of whether another Song is currently in focus.**
 - ➔ **If only “MIDI” is activated for Song, and another Song has “Play” focus, incoming MIDI will be sent to the former and MIDI sync to the latter (i.e this Song will play back), regardless of which Song is currently in focus.**

Automation Override



Automation override is activated when you manually “grab” a parameter that is being automated. If you change the setting of an automated parameter, the “Punched In” indicator lights up, and the automation data is temporarily overridden, until you either click the “Reset” button or press stop on the transport. As soon as you click Reset, the automation regains control.

See also [page 9](#).

Audio Out Clipping Indicator



All signals that are being fed into the Hardware Interface (to your audio hardware’s physical outputs) are monitored for clipping (signal overload) at the output stage.

If clipping occurs this indicator will light up, and stay lit for several seconds. If this happens, you should reduce the output level, in one of the following ways:

- ➔ **If the signals are being sent to your Hardware Interface via a Mixer, you should reduce the Master output level from the Mixer.**
This will ensure that the relative levels of the mix are kept intact. Alternatively, if the current mix doesn’t represent a “final balance”, and the clipping seems to be caused by individual channels in the mixer, you could also try reducing the output of the connected device(s), or pulling down the channel faders a bit for the “offending” channels.
- ! **Clipping can only occur in the output stage of the Hardware Interface, not in the Reason mixer or in any other Reason device. However, it is good practice to keep all mixer channel and master levels as high as possible within the normal range, for best results. For example, having to compensate channel levels by drastically reducing the Master output to avoid clipping is indicative of the mixer channel levels being set too high.**

- If the Audio Out Clipping indicator lights up, and the signals are being sent directly (not via a Mixer) to your Hardware Interface, you can check the meters in the Hardware Interface. If the red segment of any of these meters are momentarily lit, this indicates at which output(s) the clipping is occurring.

Reduce the output level of all devices connected to outputs whose meters show red.

CPU Meter



This bar graph shows the current CPU (processor) load. Note that this measures how much of the total processor power the Reason “audio engine” currently is using up. Graphics, MIDI and the “rest” of the Reason program is allotted the CPU power not used by the audio engine, so audio always has priority. See “[Optimizing Performance](#)” for more information.



REASON

9

→ Reason Hardware Interface

Introduction



The Hardware Interface is where you connect Reason with the “outside world”. This is where MIDI is received, and where audio signals are routed to ReWire channels or to the physical outputs of your audio hardware. The Hardware Interface is always present at the top of the rack, and cannot be deleted. This chapter is meant to serve as a panel reference, describing the various sections of the device. How to set up your MIDI interface and audio hardware is described in the Getting Started book and in “About Audio on Computers”.

The Hardware interface is divided into two sections: MIDI In Device and Audio Out.

MIDI In Device

Reason’s Hardware Interface can accommodate up to 64 channels of MIDI, divided into 4 buses, each with 16 MIDI channels. There are two basic ways you can route incoming MIDI to Reason devices, which is set in the Preferences - MIDI and Advanced MIDI dialogs:

→ Via the Sequencer.

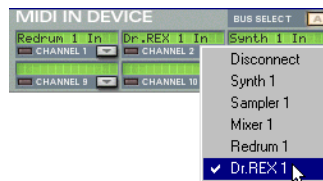
If you choose this option, the selected track’s destination device automatically receives incoming MIDI data. This means that you only have to send MIDI over the same port and channel as the sequencer is set to use (in the MIDI Preferences), to access *any* audio device in Reason. This is the easiest way of routing MIDI if you are using the built in sequencer. There are no settings you need to make in the Hardware Interface if you use the Sequencer input.

→ By using the MIDI “External Control” inputs.

This is set in the Advanced MIDI Preferences. You can select up to four buses (if your MIDI interface supports it), each with 16 MIDI channels. If this mode is used, you use the pop-up menu for each MIDI channel in the MIDI In device to select the destination device you would like to route the MIDI to. If you want to send MIDI to Reason over several channels simultaneously, you have to use the external control inputs.

Using External MIDI Control

For each MIDI channel, the MIDI In Device contains the following items:



- **The Device Pop-Up menu is used to select which device the channel should send MIDI to.**
Only existing devices are available on the menu.
- **The Name field displays the name of the device connected to the channel.**
This is blank if no device is selected.
- **A note on indicator shows if MIDI is received on this channel.**

Bus Select Buttons

These four buttons labeled A, B, C and D are used to select which of the four buses is currently displayed in the MIDI In device. If you have a multiple port interface you can use up to four buses (or ports), each with 16 MIDI channels. The Bus Select buttons determine which of the buses is currently in view in the MIDI In device.

Audio Out

Reason supports up to 64 audio output channels.

- **Each output features a meter and a green indicator which will be lit for each channel that is available.**
- ! **Remember that the Hardware Interface is where any possible audio clipping will occur in Reason. Keep an eye on the clipping indicator on the transport panel, and also on the individual meters in the Audio Out panel. If a channel pushes the meter into the red, the output level of the device should be reduced.**

Using ReWire

If you are running Reason together with a ReWire compatible host application, you can route any Reason device output to a ReWire channel by connecting the device to any of the audio inputs at the back of the Hardware Interface. In ReWire mode, all 64 channels are available and any device output routed to a ReWire channel will appear in the ReWire host application on it's own channel.

See ["Using Reason as a ReWire Slave"](#).



REASON

10

→ The Mixer

Introduction



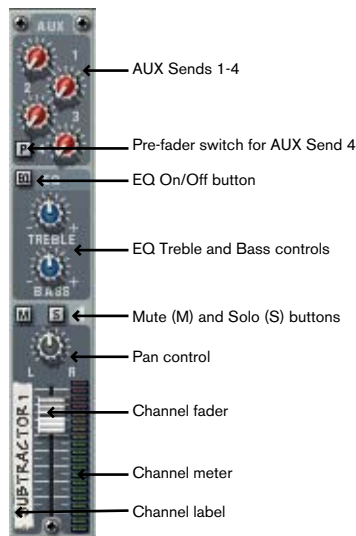
The Mixer 14:2 allows you to control the level, stereo placement (Pan), tone (EQ) and effect mix (AUX Sends) of each connected audio device.

If you have ever used a conventional hardware audio mixer, you will most likely find the Mixer very straightforward to use. It is configured with 14 (stereo) input channels, which are combined and routed to the Left and Right Master outputs. The vertical channel "strips" are identical and contain - from the top down - four Auxiliary Sends, an EQ section, Mute and Solo buttons, Pan control, and a Level fader.

Every mixer parameter can of course be automated, and should the need arise for more mixer channels, you can simply create another mixer!

! Note that if you haven't created a mixer *prior* to creating an audio device, the audio device output will be auto-routed to your audio hardware outputs via the Reason Hardware Interface (Audio Out device).

The Channel Strip



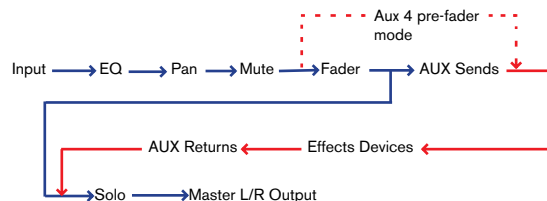
Each channel strip in the Mixer 14:2 contains the items listed on the next page:

Channel Strip Controls:

Item	Description	Value Range
Channel Fader	The channel fader is used to control the output level of each corresponding channel. By adjusting the faders, you can set the desired mix (balance) between different devices connected to the Mixer.	0 - 127
Channel Label	Each channel in the mixer that has a device connected to it, displays a read-only label with the name of the device to the left of the fader.	N/A
Channel Meter	The meter is a graphical representation of the channel output level. If the signal level pushes the meter into the range of the red area, try lowering either the output level of the device connected to the channel, or the channel fader itself, to avoid distortion.	N/A
Pan Control	Use this control to set the left/right position of the channel in the stereo field. [Command]/[Ctrl]-click the Pan knob to set Pan to the default "0" (center position).	-64 – 0 – 63
Mute (M) and Solo (S) Buttons	Clicking a channel's Mute button silences the output of that channel. Click the button again to unmute the channel. Clicking a channel's Solo button silences all other mixer channels, so that you only hear the soloed channel. Several channels can be soloed at the same time, but if this is the case, note that soloed channels can't be muted with the Mute button. To mute one of several channels in solo mode you simply "unsolo" it.	On/Off
EQ Treble and Bass controls	The EQ Treble and Bass controls is used to cut or boost the higher and lower frequencies of the signal, respectively. Click on the EQ button to activate the EQ. If you need more advanced EQ, you can always use a PEQ2 parametric EQ as an insert effect for a device. Note also the two EQ modes - see page 84 .	Treble: +/- 24 dB at 12 kHz. Bass: +/- 24 dB at 80 Hz.
Auxiliary (AUX) Effect Send 1-4	The four independent AUX Sends control the amount of channel signal that is to be sent to other devices - typically effect processors. The effect output is then normally returned to the Mixer via the AUX Return inputs (see page 84) where it is mixed with the dry (non-processed) signal. If you create an effect device when the Mixer is selected, the effect is auto-routed to the first available Send/Return connectors. You can then control the amount of effect that is to be applied to any device connected to a Mixer channel via the corresponding AUX Send knob. The AUX Send outputs are taken post channel fader, but you have the option of selecting Pre-fader mode for AUX Send 4 (by clicking the "P" button next to the send so that it lights up). In that mode, the send level is independent of the channel fader. The sends are in stereo but can be used in mono as well.	0 - 127

The Mixer signal flow

The basic signal flow for a channel in the Mixer is as follows:



Note that the Solo function is true “in-place” solo, meaning that if the channel uses Auxiliary sends routed to effect devices, the soloed output signal will also include the Aux Return signals (from the soloed channel(s) only). Hence, you will hear the soloed channel(s) including any Aux Send effects.

Note also that if the pre-fader send mode is activated for Aux 4 the send is tapped after the EQ and Pan controls but before the channel fader.

About the EQ modes



With Reason 2.5, the EQ modules in the Mixer were improved to get an even better sound and character. However, if you want to play back songs made in previous Reason versions, you may want to use the “old” EQ mode to ensure that the songs sound exactly the same.

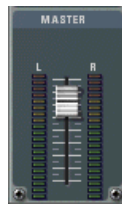
On the back of the Mixer you will find a switch for this - select “Improved EQ” for the new EQ types or “Compatible EQ” for the old-style EQ. The parameters are exactly the same in both cases.

The Auxiliary Return Section



The Auxiliary Returns provide an “extra” four stereo inputs in addition to the Mixer’s 14 stereo channels. The main function of Return channels is to provide inputs for connected Send effects devices. Each Aux Return channel has a level control, and a read-only tape label that display the name of the device connected to the Return channel.

The Master Fader



The Master L/R fader controls the summed output level of all channels in the Mixer. Use this to change the relative level of all channels, to make fade-outs etc.

Connections

All input and output connectors are as usual located on the back panel of the Mixer 14:2. Special connectors are used for “chaining” two or more Mixers together. This is described on [page 86](#).

Mixer Channel Connections

- **Each mixer channel features stereo left/right inputs for connecting audio devices.**
Use the left input when manually connecting a mono signal source.
- **In addition, there are two Control Voltage (CV) inputs (with associated voltage trim pots), for voltage controlling channel Level and Pan from other devices.**

Auxiliary (AUX) Send Out



- **There are four stereo Send Out connectors, which normally are used to connect to the inputs of effect devices.**

To connect a send to a mono-input device, use the Left (Mono) output.

When a Send is connected to an effects device, the corresponding AUX Send knob determines the level of the signal sent to the effect device for each channel. The Send Output is taken post-channel fader but you have the option of selecting pre-fader mode for AUX Send 4.

- **Note that some effects (for example the Comp-01 compressor or the PEQ2 parametric EQ) are effect types which are not designed to be used as AUX Send effects, but rather as insert effects, where the whole signal is passed through the effect.**

Alternatively, you could use AUX Send 4 in pre-fader mode and lower the channel fader completely.

Auxiliary (AUX) Returns



- **There are four stereo Return input connectors.**
These are normally connected to the left and right outputs of effect devices.

Master Left/Right Outputs



- **The Master outputs are auto-routed to the first available input pair on the Audio Hardware interface.**

This in turn sends the audio to the outputs of your audio hardware.

- ★ **Note that the Master outputs don't have to be routed directly to the Audio Hardware Interface. You could for example route the Master outputs to an effect, and then route the effect outputs to the Hardware Interface instead.**
- **In addition, there is a Control Voltage (CV) input (and an associated trim pot), for voltage controlling the Master Level from another device.**

Chaining Mixers



Two chained Mixers are connected like this, the top Mixer being the "Master" Mixer.

If you need more Mixer channels, you can simply create a new Mixer. If you do this, the Mixers are automatically connected via the "Chaining Master" and "Chaining Aux" connectors.

→ **The newly created Mixer's Master Output is connected to the original Mixer's Chaining Master input.**

The Master Out Level for the new Mixer is now controllable from the original Mixer's Master fader - so that this fader now controls the Master output level of both mixers.

→ **The newly created Mixer's four stereo Aux Send outputs is connected to the original Mixer's Chaining Aux connectors.**

The new Mixer will now have access to any Aux Send effects connected to the original Mixer, via the same corresponding Aux Send(s).

This way, the two Mixers operate as "one".

! **One exception is the Mute/Solo function, which is not chained. Thus, soloing a channel in one of the Mixers, will not mute the channels in the other Mixer.**

You can create as many Mixers as you like, they will be chained in the same way, with one Mixer remaining the "master" (i.e. it controls the Master level of all chained Mixers and supplies the Aux Send effect sources).

Partially or Non-Chained Mixers

You can also have several Mixers that are only partially or not chained at all.

- **You may for example wish to have different Aux Send effects for one Mixer.**
Then simply disconnect one or more of the Send Out to Chaining Aux connectors, and assign new Send effects.
- **You could for example send the Master output of one Mixer to another Input pair on the Audio In Hardware interface, instead of the Chaining Master inputs.**



REASON

11

→ Redrum

Introduction



At first glance, Redrum looks styled after pattern-based drum machines, like the legendary Roland 808/909 units. Indeed, it does have a row of 16 step buttons that are used for step programming patterns, just like the aforementioned classics. There are significant differences, however. Redrum features ten drum “channels” that can each be loaded with an audio file, allowing for completely open-ended sound possibilities. Don’t like the snare - just change it. Complete drum kits can be saved as Redrum Patches, allowing you to mix and match drum sounds and make up custom kits with ease.

About File Formats

Redrum reads two basic types of files:

Redrum Patches

A Redrum patch (Windows extension “.drp”) contains all settings for all ten drum sound channels, including file references to the used drum samples (but not the actual drum samples themselves). Switching patches is the same as selecting a new drum kit.

Drum Samples

Redrum can read and play back sample files of the following formats:

- Wave (.wav)
- AIFF (.aif)
- SoundFonts (.sf2)
- REX file slices (.rex2, .rex, .rcy)
- Any bit depth
- Any sample rate
- Stereo or Mono

! All samples are stored internally in 16-bit format, regardless of their original bit depth or sample rate.

Wave and AIFF are the standard audio file formats for the PC and Mac platforms, respectively. Any audio or sample editor, regardless of platform, can read and create audio files in at least one of these formats, and some of them in both formats.

SoundFonts are an open standard for wavetable synthesized audio, developed by E-mu systems and Creative Technologies.

SoundFont banks store wavetable synthesized sounds, allowing users to create and edit multi-sampled sounds in special Soundfont editing programs. The sounds can then be played back in wavetable synthesizers, typically on audio cards. The samples in a SoundFont are stored hierarchically in different categories: User Samples, Instruments, Presets etc. The Redrum allows you to browse and load single SoundFont samples, *not* entire soundfonts.

REX files are files created in ReCycle – a program designed for working with sampled loops. It works by “slicing” up a loop and making separate samples of each beat, which makes it possible to change the tempo of loops without affecting the pitch and to edit the loop as if it was built up of individual sounds. The Redrum lets you browse REX files and load separate slices from it as individual samples.

Using Patches

When you create a new Redrum device it is empty. Before it can play back any audio you must first load a Redrum patch (or create one from scratch, by loading individual drum samples). A Redrum patch contains settings for the ten drum sound channels, complete with file references to the drum samples used.

! **Redrum patterns are *not* part of the patch!**

Loading a Patch

To load a patch, use one of the following methods:

- **Use the browser to locate and open the desired patch.**

To open the browser, select "Browse Redrum Patches" from the Edit menu or device context menu, or click the folder button in the patch section on the device panel.



- **Once you have selected a patch, you can step between all the patches in the same folder by using the arrow buttons next to the patch name display.**
- **If you click on the patch name display on the device panel, a pop-up menu will appear, listing all patches in the current folder.**
This allows you to quickly select another patch in the same folder, without having to step through each one in turn.

Checking the Sounds in a Patch

There are two ways you can listen to the sounds in a patch without programming a pattern:

- **By clicking the Trigger (arrow) button at the top of each drum sound channel.**



- **By playing the keys C1 to A1 on your MIDI keyboard.**

C1 plays drum sound channel 1 and so on. See also [page 97](#).

Both these methods play back the drum sample for the corresponding drum sound channel, with all settings for the sound applied.

Creating a new Patch

To create a patch of your own (or modify an existing patch), you use the following basic steps:

1. **Click the folder button for a drum sound channel.**

The Redrum sample browser opens.



2. **Locate and open a drum sample.**

You will find a large number of drum samples in the Factory Sound Bank (in the folder Redrum Drum Kits/xclusive drums-sorted). You can also use any AIFF, Wave, SoundFont sample or REX file slice for this.

3. **Make the desired settings for the drum sound channel.**

The parameters are described on [page 93](#).

4. **Repeat steps 1 and 3 for the other drum sound channels.**

5. **When you're satisfied with the drum kit, you can save the patch by clicking the Floppy Disk button in the patch section on the device panel.**

Note however, that you don't necessarily need to save the patch - all settings are included when you save the song.

Loading REX file slices

Loading slices from within a REX file is done much in the same way as loading "regular" samples:

1. **Open the sample browser as described above.**

2. **Browse to a REX file.**

Possible extensions are ".rex2", ".rex" and ".rcy".

3. **Select the file and click "Open".**

The browser will now display a list of all the separate slices within the REX file.

4. **Select the desired slice and click open.**

The slice is loaded into the Redrum.

Creating an Empty Patch

To “initialize” the settings in the Redrum, select “Initialize Patch” from the Edit menu or the device context menu. This removes all samples for all drum sound channels, and sets all parameters to their default values.

Programming Patterns

About Pattern Selection

As described in the Getting Started book, each pattern device (such as the Redrum) has 32 pattern memories, divided into four banks. To select a pattern, click a Pattern button (or, if the desired pattern is in another bank, first click the Bank button and then click the Pattern button).

- **If you select a new pattern during playback, the change will take effect on the next downbeat (according to the time signature set in the transport panel).**

If you automate pattern changes in the main sequencer, you can make them happen at any position - see [page 29](#).

- **Note that you cannot load or save patterns - they are only stored as part of a song.**

However, you can move patterns from one location to another (even between songs) by using the Cut, Copy and Paste Pattern commands. This is explained in the chapter “Using Pattern Devices” in the Getting Started book.

Pattern Programming Basics

If you are unfamiliar with step programming patterns, the basic principle is very intuitive and easy to learn. Proceed as follows:

- 1. Load a Redrum patch, if one isn't already loaded.**
- 2. Make sure an empty pattern is selected.**
If you like, use the Clear Pattern command on the Edit menu or device context menu to make sure.
- 3. Make sure that the “Enable Pattern Section” and the “Pattern” buttons are activated (lit).**



- 4. Press the “Run” button.**

There will be no sound, as no pattern steps have been recorded yet. But as you can see, the LEDs over the Step button light up consecutively, moving from left to right, and then starts over. Each Step button represents one “step” in the Pattern.

5. **Select a Redrum channel, by clicking the “Select” button at the bottom of the channel.**

The button lights up, indicating that this channel and the drum sound it contains is selected.



6. **While in Run mode, press Step button 1, so that it lights up.**
The selected sound will now play every time Step 1 is “passed over”.
7. **Clicking other Step buttons so they light up will play back the selected sound as the sequencer passes those steps.**
Clicking on a selected (lit) step button a second time removes the sound from that step and the button goes dark again. You can click and drag to add or remove steps quickly.
8. **Select another Redrum channel to program steps for that sound.**
Selecting a new sound or channel also removes the visual indications (static lit buttons) of step entries for the previously selected sound. The step buttons always show step entries for the currently selected sound.
9. **Continue switching between sounds, and programming steps to build your pattern.**
Note that you can erase or add step entries even if Run mode isn’t activated.

Setting Pattern Length

You may want to make settings for Pattern length, i.e the number of steps the pattern should play before repeating:



- **Use the “Steps” spin controls to set the number of steps you wish the pattern to play.**
The range is 1 to 64. You can always extend the number of steps at a later stage, as this will merely add empty steps at the end of the original pattern. You could also make it shorter, but that would (obviously) mean that the steps “outside” the new length won’t be heard. These steps aren’t erased though; if you raise the Steps value again, the steps will be played back again.

About the “Edit Steps” Switch

If you set the pattern length to more than 16 steps, the pattern steps following after the 16th won’t be visible, although they will play back. To view and be able to edit the next 16 steps, you have to set the Edit Steps switch to 17-32. To see and edit steps beyond 32 you set the switch to 33-48, and so on.

Setting Pattern Resolution



Redrum always follows the tempo setting on the transport panel, but you can also make Redrum play in different “resolutions” in relation to the tempo setting. Changing the Resolution setting changes the length of each step, and thereby the “speed” of the pattern.

This is explained in the chapter “Using Pattern Devices” in the Getting Started book.

Step Dynamics

When you enter step notes for a drum sound, you can set the velocity value for each step to one of three values: Hard, Medium or Soft. This is done by setting the Dynamic switch before entering the note.



The color of the step buttons reflect the dynamics for each step. Soft notes are light yellow, Medium notes are orange and Hard are red.

- **When the Medium value is selected, you can enter Hard notes by holding down [Shift] and clicking.**
In the same way, you can enter Soft notes by holding down [Option] (Mac) or [Alt] (Windows) and clicking. Note that this doesn’t change the Dynamic setting on the device panel - it only affects the notes you enter.

- **When you use different dynamics, the resulting difference in the sound (loudness, pitch, etc.), is governed by the “VEL” knob settings for each drum channel (see [page 93](#)).**
If no velocity amount is set for a drum channel, it will play back the same, regardless of the Dynamic setting.
- **To change the dynamics for an already programmed step, set the switch to the dynamic value you wish to change it to and click on the step.**
- ! **Note that if you are triggering Redrum via MIDI or from the main sequencer, the sounds will react to velocity like any other audio device. The Dynamic values are there to offer velocity control when using the built-in pattern sequencer.**

Pattern Shuffle

Shuffle is a rhythmic feature, that gives the music a more or less pronounced swing feel. It works by delaying all sixteenth notes that fall in between the eighth notes.

You can activate or deactivate shuffle individually for each Redrum pattern by clicking the Shuffle button on the device panel.



However, the amount of shuffle is set globally with the Pattern Shuffle control on the transport panel.



Flam

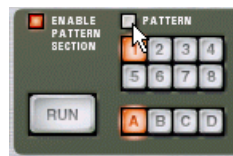


A flam is when you double-strike a drum, to create a rhythmic or dynamic effect. Applying flam to a step entry will add a second “hit” to a drum sound. The flam amount knob determines the delay between the two hits.

To add a flam drum note, proceed as follows:

- 1. Activate flam by clicking the Flam button.**
 - 2. Click on a step to add a note (taking the Dynamic setting into account as usual).**
A red LED is lit above the step to indicate that flam will be applied to that step.
 - 3. Use the Flam knob to set the desired amount of flam.**
The flam amount is global for all patterns in the device.
- **To add or remove flam to or from an existing step note, click directly on the corresponding flam LED.**
You can also click and drag on the LEDs to add or remove several flam steps quickly.
 - **Applying flam to several consecutive step entries is a quick way to produce drum rolls.**
By adjusting the Flam knob you can create 1/32 notes even if the step resolution is 1/16, for example.

The Pattern Enable switch



If you deactivate the “Pattern” button, the pattern playback will be muted, starting at the next downbeat (exactly as if you had selected an empty (silent) pattern). For example, this can be used for bringing different pattern devices in and out of the mix during playback.

The Enable Pattern Section switch

If this is deactivated, Redrum will function as a pure “sound module”, i.e. the internal Pattern sequencer is disengaged. Use this mode if you wish to control Redrum exclusively from the main sequencer or via MIDI (see [page 97](#)).

Pattern Functions

When a Redrum device is selected, you will find some specific pattern functions on the Edit menu (and on the device context menu):

Function	Description
Shift Pattern Left/Right	These functions move all notes in the pattern one step to the left or right.
Shift Drum Left/Right	The Shift Drum functions move all notes for the selected drum channel (the channel for which the Select button is lit) one step to the left or right.
Randomize Pattern	Creates a random pattern. Random patterns can be great starting points and help you get new ideas.
Randomize Drum	Creates a random pattern for the selected drum sound only - the notes for the other drum sound channels are unaffected.
Alter Pattern	The Alter Pattern function modifies the selected pattern by "shuffling" the current pattern notes and redistributing them among the drum sounds at random. This creates a less chaotic pattern than the "Randomize Pattern" function. Note that there must be something in the pattern for the function to work on - using an Alter function on an empty pattern will not do anything.
Alter Drum	Works like the "Alter Pattern" function, but affects the selected drum sound only.

Chaining Patterns

When you have created several patterns that belong together, you most probably want to make these play back in a certain order. This is done by recording or inserting pattern changes into the main sequencer. See [page 29](#).

Converting Pattern Data to Notes

You can convert Redrum Patterns to notes in the main sequencer. This allows you to edit the notes freely, create variations or use Groove quantizing. This is described on [page 12](#).

Redrum Parameters

Drum Sound Settings

Redrum features ten drum sound channels that can each be loaded with a Wave or AIFF sample or a sample from a SoundFont bank. Although they are basically similar, there are three "types" of drum sound channels, with slightly different features. This makes some channels more suitable for certain types of drum sounds, but you are of course free to configure your drum kits as you like.

On the following pages, all parameters will be listed. If a parameter is available for certain drum sound channels only, this will be stated.

Mute & Solo



At the top of each drum sound channel, you will find a Mute (M) and a Solo (S) button. Muting a channel silences its output, while Soloing a channel mutes all other channels. Several channels can be muted or soloed at the same time.

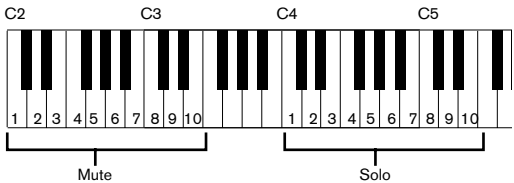
You can also use keys on your MIDI keyboard to mute or solo individual drum sounds in real time.

→ **The keys C2 to E3 (white keys only) will mute individual drum channels starting with channel 1.**

The sounds are muted for as long as you hold the key(s) down.

→ **The keys C4 to E5 (white keys only) will solo individual drum channel, starting with channel 1.**

The sounds are soloed for as long as you hold the key(s) down.



This is a great way to bring drum sounds in and out of the mix when playing Reason live. You can also record the drum channel Mutes in the main sequencer, just like any other controller (see [page 22](#)).

The Effect Sends (S1 & S2)



On the back panel of Redrum you will note two audio connections labeled “Send Out” 1 and 2. When you create a Redrum device, these will by default be auto-routed to the first two “Chaining Aux” inputs on the Mixer device (provided that these inputs aren’t already in use).

This feature allows you to add effects to independent drum sounds in the Redrum.

- **Raising the S1 knob for a drum sound channel will send the sound to the first send effect connected to the mixer.**
Similarly, the S2 knob governs the send level to the second send effect in the mixer.
- **Note that there must be send effects connected to the AUX Sends and Returns in the mixer for this to work.**
- **Also note that if Redrum is soloed in the Mixer the effect sends will be muted.**
- **Another way to add independent effects to drum sounds is to use the independent drum outputs.**
See [page 97](#).

Pan



Sets the Pan (stereo position) for the channel.

- **If the LED above the Pan control is lit, the drum sound uses a stereo sample.**
In that case, the Pan control serves as a stereo balance control.

Level and Velocity



The Level knob sets the volume for the channel. However, the volume can also be affected by velocity (as set with the Dynamic value, or as played via MIDI). How much the volume should be affected by velocity is set with the “Vel” knob.

- **If the Vel knob is set to a positive value, the volume will become louder with increasing velocity values.**
The higher the Vel value, the larger the difference in volume between low and high velocity values.
- **A negative value inverts this relationship, so that the volume decreases with higher velocity values.**
- **If the Vel knob is set to zero (middle position), the sound will play at a constant volume, regardless of the velocity.**
When Vel is set to zero, the LED above the knob goes dark.

Length and the Decay/Gate switch



The Length knob determines the length of the drum sound, but the result depends on the setting of the Decay/Gate switch:

- **In Decay mode (switch down), the sound will decay (gradually fade out) after being triggered. The decay time is determined by the Length setting.**
In this mode, it doesn’t matter for how long a drum note is held (if played back from the main sequencer or via MIDI) - the sound will play the same length for short notes as for long notes. This is the traditional “drum machine” mode.

- **In Gate mode (switch up), the sound will play for the set Length, and then be cut off.**

Furthermore, if a sound set to Gate mode is played from the main sequencer, from a CV/Gate device or via MIDI, the sound will be cut off when the note ends or after the set Length, depending on which comes first. Or in other words, the sound plays for as long as you hold the note, but the Length setting serves as the maximum length for the sound.

There are several uses for the Gate mode:

- For “gated” drum sounds, when the tail of the sound is abruptly cut off as an effect.
- For when you want to use very short sounds, and don’t want them to “lose power” by being faded out.
- For when you play the Redrum from the sequencer or via MIDI, with sounds for which the length is important (e.g. when using the Redrum as a sound effects module).

- ! **Audio samples sometimes contain a “loop”, which is set by editing the audio in a sample editor. This loop repeats a part of the sample to produce sustain as long as you hold down a note. Drum samples usually don’t contain loops, but who is to say that Redrum should only play drum samples?**

Note that if a sample contains a loop, and Length is set to maximum, the sound will have infinite sustain, in other words it will never become silent, even if you stop playback. Decreasing the Length setting solves this problem.

Pitch



Sets the pitch of the sound. The range is +/- 1 octave.

- **When the pitch is set to any other value than 0, the LED above the knob lights up to indicate that the sample isn’t played back at its original pitch.**

Pitch Bend



By setting the Bend knob to a positive or negative value, you specify the start pitch of the sound (relative to the Pitch setting). The pitch of the sound will then be bent to the main Pitch value. Thus, selecting a positive Bend value will cause the pitch to start higher and bend down to the original Pitch, and vice versa.

- **The Rate knob determines the bend time - the higher the value, the slower the bend.**
- **The Vel knob determines how the Bend amount should be affected by velocity.**
With a positive Vel value, higher velocity results in wider pitch bends.
- **The Bend and Vel knobs have LEDs that light up when the functions are activated (i.e. when a value other than zero is selected).**
- ! **Pitch bend is available for drum sound channels 6 and 7 only.**

Tone



The Tone knob determines the brightness of the drum sound. Raising this parameter results in a brighter sound. The Vel knob determines whether the sound should become brighter (positive Vel value) or darker (negative Vel value) with higher velocity.

- **The Tone and Vel knobs have LEDs that light up when the functions are activated (i.e. when a value other than zero is selected).**
- ! **The Tone controls are available for drum sound channels 1, 2 and 10 only.**

Sample Start



The Start parameter allows you to adjust the start point of the sample. The higher the Start value, the further the start point is moved “into” the sample. If you set the Start Velocity knob to a positive amount, the sample start point is moved forward with higher velocities. A negative Start Velocity amount inverts this relationship.

- **When Start Velocity is set to any other value than zero, the LED above the knob lights up.**
- **A negative Start Velocity amount is only useful if you have set the Start parameter to a value higher than 0.**
By raising the Start value a bit and setting Start Velocity to a negative value, you can create rather realistic velocity control over some drum sounds. This is because the very first transients in the drum sound will only be heard when you play hard notes.
- ! **The Sample Start settings are available for drum sound channels 3-5, 8 and 9.**

Global Settings

Channel 8 & 9 Exclusive



If this button is activated, the sounds loaded into drum channels 8 and 9 will be exclusive. In other words, if a sound is played in channel 8 it will be silenced the moment a sound is triggered in channel 9, and vice versa.

The most obvious application for this feature is to “cut off” an open hi-hat with a closed hi-hat, just like a real one does.

High Quality Interpolation



When this is activated, the sample playback is calculated using a more advanced interpolation algorithm. This results in better audio quality, especially for drum samples with a lot of high frequency content.

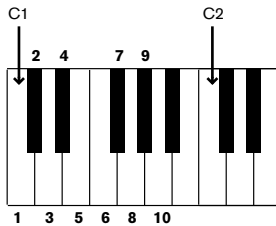
- **High Quality Interpolation uses more computer power - if you don't need it, it's a good idea to turn it off!**
Listen to the drum sounds in a context and determine whether you think this setting makes any difference.
- ! **If you are using a Macintosh with a G4 (AltiVec) processor, turning High Quality Interpolation off makes no difference.**

Master Level

The Master Level knob in the top left corner of the device panel governs the overall volume from Redrum.

Using Redrum as a Sound Module

The drum sounds in Redrum can be played via MIDI notes. Each drum sound is triggered by a specific note number, starting at C1 (MIDI note number 36):



This allows you to play Redrum live from a MIDI keyboard or a MIDI percussion controller, or to record or draw drum notes in the main sequencer. If you like, you can combine pattern playback with additional drum notes, such as fills and variations. However:

! If you want to use Redrum purely as a sound module (i.e. without pattern playback) you should make sure that the “Enable Pattern Section” button is deactivated. Otherwise, the Redrum pattern sequencer will start as soon as you start the main sequencer.

Connections



On the back of the Redrum you will find the following connections:

For each drum sound channel:

Connection	Description
Audio Outputs	There are individual audio outputs for each drum sound channel, allowing you to route a drum sound to a separate channel in the mixer, possibly via insert effects, etc. For mono sounds, use the “Left (Mono)” output (and pan the sound using the Pan control in the mixer). When you use an individual output for a sound, the sound is automatically excluded from the master stereo output.
Gate Out	This sends out a gate signal when the drum sound is played (from a pattern, via MIDI or by using the Trigger button on the device panel). This lets you use the Redrum as a “trig sequencer”, controlling other devices. The length of the gate signal depends on the Decay/Gate setting for the sound: In Decay mode, a short “trig pulse” is sent out, while in Gate mode, the gate signal will have the same length as the drum note (see page 94).
Gate In	Allows you to trigger the sound from another CV/Gate device. All settings apply, just as when playing the drum sound conventionally.
Pitch CV In	Lets you control the pitch of the drum sound from another CV device.

Others

Connection	Description
Send Out 1-2	Outputs for the send signals controlled with the S1 and S2 knobs, as described on page 94 .
Stereo Out	This is the master stereo output, outputting a mix of all drum sounds (except those for which you use individual outputs).



REASON

12

→ Subtractor Synthesizer

Introduction

Subtractor is an analog-type polyphonic synthesizer based on subtractive synthesis, the method used in analog synthesizers. This chapter will go through all parameters of each section of Subtractor. In addition to the parameter descriptions, the chapter also includes a few tips and tricks to help you get the most out of the Subtractor synthesizer.

! It is recommended that you start with default settings (an “Init Patch”) if you intend to follow the examples in this chapter, unless otherwise is stated. An Init Patch is created by selecting “Initialize Patch” from the Edit menu. If you wish to keep the current settings, save them before initializing.

The Subtractor has the following basic features:

- **Up to 99 Voice Polyphony.**
You can set the number of voices for each Patch.
- **Dual Filters.**
A combination of a multimode filter and a second, linkable, lowpass filter allows for complex filtering effects. See [page 105](#).
- **Two Oscillators, each with 32 waveforms.**
See [page 100](#).
- **Frequency Modulation (FM).**
See [page 104](#).
- **Oscillator Phase Offset Modulation.**
This is a unique Subtractor feature that generates waveform variations. See [page 103](#).
- **Two Low Frequency Oscillators (LFO:s)**
See [page 111](#).
- **Three Envelope Generators.**
See [page 109](#).
- **Extensive Velocity Control.**
See [page 113](#).
- **Extensive CV/Gate Modulation possibilities.**
See [page 116](#).
- ! Loading and saving Patches is described in the chapter “Working with Patches” in the Getting Started book.**

The Oscillator Section



Subtractor provides two oscillators. Oscillators are the main sound generators in Subtractor, the other features are used to *shape* the sound of the oscillators. Oscillators generate two basic properties, waveform and pitch (frequency). The type of waveform the oscillator produces determines the harmonic content of the sound, which in turn affects the resultant sound quality (timbre). Selecting an oscillator waveform is usually the starting point when creating a new Subtractor Patch from scratch.

Oscillator 1 Waveform



Oscillator 1 provides 32 waveforms. The first four are standard waveforms, and the rest are “special” waveforms, some of which are suitable for emulating various musical instrument sounds.





✦ **It is worth noting here that all waveforms can be radically transformed using Phase offset modulation (see [page 103](#)).**

→ **To select a waveform, click the spin controls to the right of the “Waveform” LED display.**

The first 4 basic waveforms are shown as standard symbols, and the special waveforms are numbered 5 - 32.

Here follows a brief description of the Subtractor waveforms:

! Please note that the descriptions of the waveforms sound or timbre is merely meant to provide a basic guideline, and shouldn't be taken too literally. Given the myriad ways you can modulate and distort a waveform in Subtractor, you can produce extremely different results from any given waveform.

Waveform	Description
 Sawtooth	This waveform contains all harmonics and produces a bright and rich sound. The Sawtooth is perhaps the most "general purpose" of all the available waveforms.
 Square	A square wave only contains odd number harmonics, which produces a distinct, hollow sound.
 Triangle	The Triangle waveform generates only a few harmonics, spaced at odd harmonic numbers. This produces a flute-like sound, with a slightly hollow character.
 Sine	The sine wave is the simplest possible waveform, with no harmonics (overtones). The sine wave produces a neutral, soft timbre.
5	This waveform emphasizes the higher harmonics, a bit like a sawtooth wave, only slightly less bright-sounding.
6	This waveform features a rich, complex harmonic structure, suitable for emulating the sound of an acoustic piano.
7	This waveform generates a glassy, smooth timbre. Good for electric piano-type sounds.
8	This waveform is suitable for keyboard-type sounds such as harpsichord or clavinet.
9	This waveform is suitable for electric bass-type sounds.
10	This is a good waveform for deep, sub-bass sounds.
11	This produces a waveform with strong formants, suitable for voice-like sounds.
12	This waveform produces a metallic timbre, suitable for a variety of sounds.
13	This produces a waveform suitable for organ-type sounds.
14	This waveform is also good for organ-type sounds. Has a brighter sound compared to waveform 13.

Waveform	Description
15	This waveform is suitable for bowed string sounds, like violin or cello.
16	Similar to 15, but with a slightly different character.
17	Another waveform suitable for string-type sounds.
18	This waveform is rich in harmonics and suitable for steel string guitar-type sounds.
19	This waveform is suitable for brass-type sounds.
20	This waveform is suitable for muted brass-type sounds.
21	This waveform is suitable for saxophone-like sounds.
22	A waveform suitable for brass and trumpet-type sounds.
23	This waveform is good for emulating mallet instruments such as marimba.
24	Similar to 23, but with a slightly different character.
25	This waveform is suitable for guitar-type sounds.
26	This is a good waveform for plucked string sounds, like harp.
27	Another waveform suitable for mallet-type sounds (see 23-24), but has a brighter quality, good for vibraphone-type sounds.
28	Similar to 27, but with a slightly different character.
29	This waveform has complex, enharmonic overtones, suitable for metallic bell-type sounds.
30	Similar to 29, but with a slightly different character. By using FM (see page 104) and setting the Osc Mix to Osc 1, this and the following two waveforms can produce noise.
31	Similar to 30, but with a slightly different character.
32	Similar to 30, but with a slightly different character.

Setting Oscillator 1 Frequency - Octave/ Semitone/Cent



By clicking the corresponding up/down buttons you can tune, i.e. change the frequency of Oscillator 1 in three ways:

→ In Octave steps

The range is 0 - 9. The default setting is 4 (where "A" above middle "C" on your keyboard generates 440 Hz).

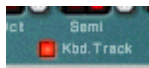
→ In Semitone steps

Allows you to raise the frequency in 12 semitone steps (1 octave).

→ In Cent steps (100th of a semitone)

The range is -50 to 50 (down or up half a semitone).

Oscillator Keyboard Tracking



Oscillator 1 has a button named "Kbd. Track". If this is switched off, the oscillator pitch will remain constant, regardless of any incoming note pitch messages, although the oscillator still reacts to note on/off messages. This can be useful for certain applications:

→ When Frequency Modulation (FM - see [page 104](#)) or Ring Modulation (see [page 105](#)) is used.

This produces enharmonic sounds with very varying timbre across the keyboard.

→ For special effects and non-pitched sounds (like drums or percussion) that should sound the same across the keyboard.

Using Oscillator 2



You activate Osc 2 by clicking the button next to the text "Osc 2". Setting oscillator frequency and keyboard tracking is identical to Oscillator 1.

Adding a second oscillator enables many new modulation possibilities which can produce richer timbres. A basic example is to slightly detune (+/- a few cents) one of the oscillators. This slight frequency offset causes the oscillators to "beat" against each other, producing a wider and richer sound. Also, by combining two different waveforms, and adding frequency or ring modulation, many new timbres can be created.

Oscillator Mix



The Osc Mix knob determines the output balance between Osc 1 and Osc 2. To be able to clearly hear both oscillators, the "Osc Mix" knob should be set somewhere around the center position. If you turn the Mix knob fully to the left, only Osc 1 will be heard, and vice versa. [Command]/[Ctrl]-clicking the knob sets the Mix parameter to center position.

Oscillator 2 Waveform

The waveform alternatives for Oscillator 2 are identical to those of Oscillator 1.

However, the Noise Generator provides a third sound generating source (in addition to the two oscillators) in Subtractor, and could be regarded as an "extra" waveform for Oscillator 2, as it is internally routed to the Oscillator 2 output. See below for a description of the Noise Generator.

Noise Generator



The Noise Generator could be viewed as an oscillator that produces noise instead of a pitched waveform. Noise can be used to produce a variety of sounds, the classic example being “wind” or “rolling wave” sounds, where noise is passed through a filter while modulating the filter frequency. Other common applications include non-pitched sounds like drums and percussion, or simulating breath noises for wind instruments. To use the Noise Generator, select an Init Patch and proceed as follows:

- 1. **Turn Osc 2 off.**
 - 2. **Click the button (in the Noise Generator section) to activate the Noise Generator.**
If you play a few notes on your MIDI instrument you should now hear Osc1 mixed with the sound of the Noise Generator.
 - 3. **Turn the Mix knob fully to the right, and play a few more notes.**
Now just the Noise Generator will be heard.
- **Thus, the output of the Noise Generator is internally routed to Osc 2.**
If you switch Osc 2 on, the noise will be mixed with the Osc 2 waveform.

There are three Noise Generator parameters. These are as follows:

Parameter	Description
Noise Decay	This controls how long it takes for the noise to fade out when you play a note. Note that this is independent from the Amp Envelope Decay parameter (see page 110), allowing you to mix a short “burst” of noise at the very beginning of a sound, i.e. a pitched sound that uses oscillators together with noise.
Noise Color	This parameter allows you to vary the character of the noise. If the knob is turned fully clockwise, pure or “white” noise (where all frequencies are represented with equal energy) is generated. Turning the knob anti-clockwise produces a gradually less bright sounding noise. Fully anti-clockwise the noise produced is an earthquake-like low frequency rumble.
Level	Controls the level of the Noise Generator.

Phase Offset Modulation

A unique feature of the Subtractor oscillators is the ability to create an extra waveform within one oscillator, to offset the *phase* of that extra waveform, and to modulate this phase offset. By subtracting or multiplying a waveform with a phase offset copy of itself, very complex waveforms can be created. Sounds complicated? Well, the theory behind it might be, but from a user perspective it is just a method of generating new waveforms from existing waveforms.

A seasoned synth programmer using Subtractor for the first time may wonder why the Subtractor oscillators (seemingly) cannot provide the commonly used pulse waveform and the associated pulse width modulation (PWM). Or oscillator sync, another common feature in analog synthesizers. The simple answer is that Subtractor can easily create pulse waveforms (with PWM) and oscillator sync-sounds, and a lot more besides, partly by the use of phase offset modulation.

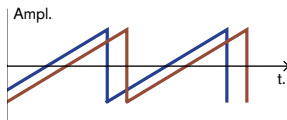


Each oscillator has it's own Phase knob and a selector button. The Phase knob is used to set the amount of phase offset, and the selector switches between three modes:

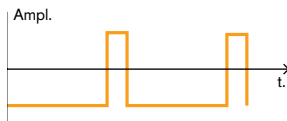
- Waveform multiplication (x)
- Waveform subtraction (-)
- No phase offset modulation (o).

When phase offset modulation is activated, the oscillator creates a *second* waveform of the same shape and offsets it by the amount set with the Phase knob. Depending on the selected mode, Subtractor then either *subtracts* or *multiplies* the two waveforms with each other. The resulting waveforms can be seen in the illustration below.

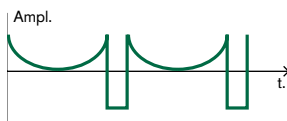
1. The two offset waveforms:



2. The result of subtraction:



3. The result of multiplication:



- In example 1, we see two sawtooth waves with a slight offset.
- Example 2 shows that *subtracting* one slightly offset sawtooth wave from the other, produces a pulse wave. If you modulate the Phase offset parameter (with for example an LFO), the result will be pulse width modulation (PWM).
- Example 3 shows the resulting waveform when *multiplying* the offset waves with each other. As you can see (and hear if you try it), multiplying waveforms can produce very dramatic and sometimes unexpected results.

Using phase offset modulation can create very rich and varied timbres, especially when used along with LFO or Envelopes to modulate the phase offset.

🌟 To get a “feel” for this concept, you could study Patches that use phase offset modulation, and maybe tweak some of the Phase Offset parameters to find out what happens. Try “SyncedUp” in the Polysynth category in the Factory Soundbank for an example of osc sync or “Sweeping Strings” (in the Pads category) for an example of PWM.

! **Note that if you activate waveform *subtraction* with a Phase offset set to “0” for an oscillator, the second waveform will cancel out the original waveform completely, and the oscillator output will be silent. If you set the Phase Offset knob to any other value than zero, the sound returns.**

Frequency Modulation (FM)



In synthesizer-speak, Frequency Modulation, or FM, is when the frequency of one oscillator (called the “carrier”) is modulated by the frequency of another oscillator (called the “modulator”). Using FM can produce a wide range of harmonic and non harmonic sounds. In Subtractor, Osc 1 is the carrier and Osc 2 the modulator. To try out some of the effects FM can produce, proceed as follows:

1. **Select an Init Patch by selecting “Initialize Patch” from the Edit menu.**
 2. **Activate Osc 2.**
As you need both a carrier and a modulator to produce FM, turning the FM knob will not produce any effect unless you first activate Osc 2. For classic FM sounds, use sine wave on oscillator 1 and triangle wave on oscillator 2.
 3. **Use the FM knob to set the FM amount to a value of about 50.**
As you can hear, the timbre changes, but the effect isn’t very pronounced yet.
 4. **Turn the Osc Mix knob fully to the left, so that only the sound of Osc 1 is heard.**
The modulator (Osc 2) still affects Osc 1, even though the Osc 2 output is muted.
 5. **Now, hold down a note on your MIDI keyboard and tune Osc 2 a fifth up from the original pitch by setting the Osc 2 frequency “Semi” parameter to a value of 7.**
As you can hear, for each semitone step you vary the Osc 2 frequency, the timbre changes dramatically. Setting Osc 2 frequency to certain musical intervals (i.e. fourth, fifth or octave semitone steps) produces harmonic, rich timbres, almost like tube distortion. Setting Osc 2 to non-musical intervals usually results in complex, enharmonic timbres.
- 🌟 Experiment with different oscillator parameters such as phase offset modulation, changing the waveforms etc. and listen to how they affect the sound of frequency modulation.

Using the Noise Generator as the Modulator source

As explained earlier, the Noise Generator is internally routed to the Osc 2 output. Hence, if you deactivate Osc 2, and activate the Noise Generator while using FM, the noise will be used to frequency modulate Osc 1.



SUBTRACTOR SYNTHESIZER

- ✦ With the Noise Generators default settings, this will sound much like colored noise. But by changing (lowering) the Noise Generator Decay parameter, so that the noise modulates only the attack portion of the sound can produce more interesting results. You could also use a combination of noise and Osc 2.

Ring Modulation



Ring Modulators basically multiply two audio signals together. The ring modulated output contains added frequencies generated by the sum of, and the difference between, the frequencies of the two signals. In the Subtractor Ring Modulator, Osc 1 is multiplied with Osc 2 to produce sum and difference frequencies. Ring modulation can be used to create complex and enharmonic, bell-like sounds.

- 1. Select an Init Patch by selecting “Initialize Patch” from the Edit menu.**
Save any current settings you wish to keep before initializing.
- 2. Activate Ring Modulation with the button in the lower right corner of the oscillator section.**
- 3. Activate Osc 2.**
You need to activate Osc 2 before any ring modulation can happen.
- 4. Turn the Osc Mix knob fully to the right, so that only the sound of Osc 2 is heard.**
Osc 2 provides the ring modulated output.
- 5. If you play a few notes while varying the frequency of either oscillator, by using the Semitone spin controls, you can hear that the timbre changes dramatically.**
If the oscillators are tuned to the same frequency, and no modulation is applied to either the Osc 1 or 2 frequency, the Ring Modulator won't do much. It is when the frequencies of Osc 1 and Osc 2 *differ*, that you get the “true” sound of ring modulation.

The Filter Section



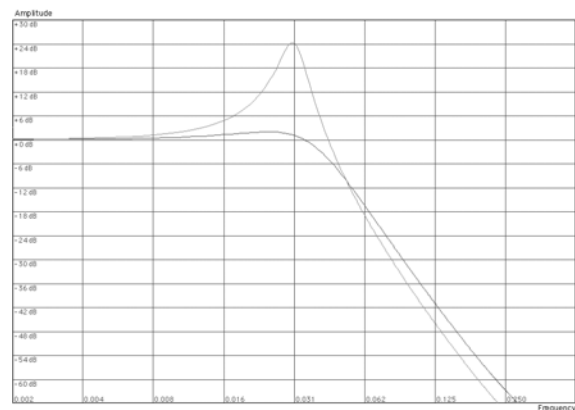
In subtractive synthesis, a filter is the most important tool for shaping the overall timbre of the sound. The filter section in Subtractor contains two filters, the first being a multimode filter with five filter types, and the second being a low-pass filter. The combination of a multimode filter and a lowpass filter can be used to create very complex filter effects.

Filter 1 Type

With this multi-selector you can set Filter 1 to operate as one of five different types of filter. The five types are illustrated and explained on the following pages:

→ 24 dB Lowpass (LP 24)

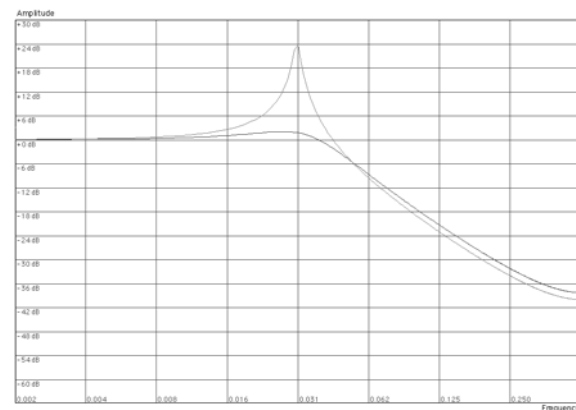
Lowpass filters let low frequencies pass and cut out the high frequencies. This filter type has a fairly steep roll-off curve (24dB/Octave). Many classic synthesizers (Minimoog/Prophet 5 etc.) use this filter type.



The darker curve illustrates the roll-off curve of the 24dB Lowpass Filter. The lighter curve in the middle represents the filter characteristic when the Resonance parameter is raised.

→ 12 dB Lowpass (LP 12)

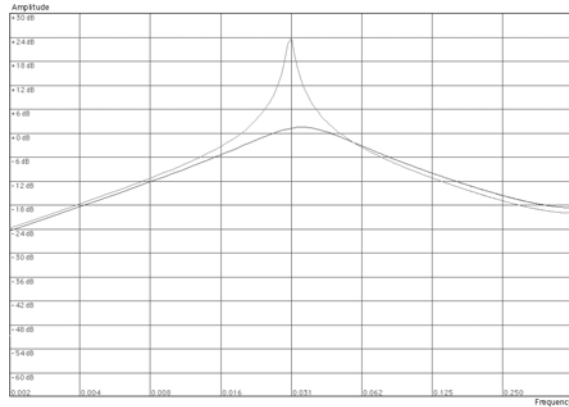
This type of lowpass filter is also widely used in analog synthesizers (Oberheim, early Korg synths etc.). It has a gentler slope (12 dB/Octave), leaving more of the harmonics in the filtered sound compared to the LP 24 filter.



The darker curve illustrates the roll-off curve of the 12dB Lowpass Filter. The lighter curve in the middle represents the filter characteristic when the Resonance parameter is raised.

→ Bandpass (BP 12)

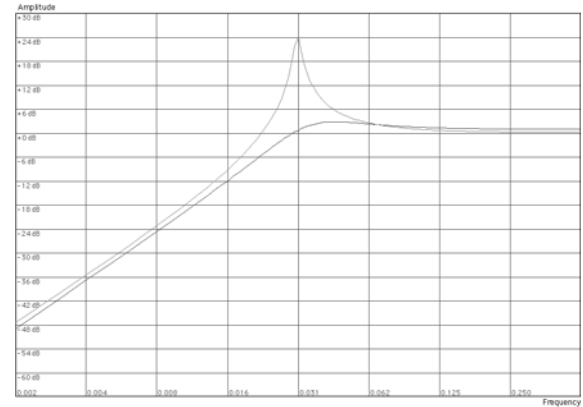
A bandpass filter cuts both high and low frequencies, while midrange frequencies are not affected. Each slope in this filter type has a 12 dB/Octave roll-off.



The darker curve illustrates the roll-off curve of the Bandpass Filter. The lighter curve in the middle represents the filter characteristic when the Resonance parameter is raised.

→ Highpass (HP12)

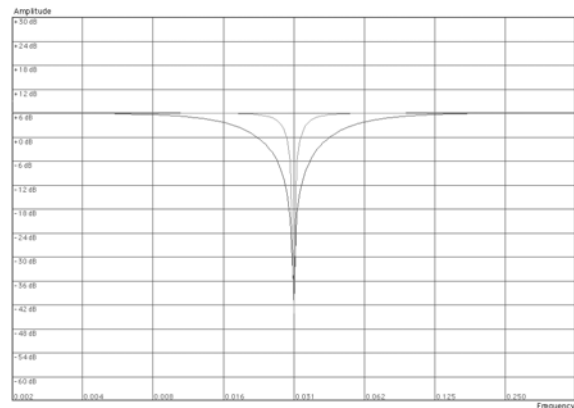
A highpass filter is the opposite of a lowpass filter, cutting out the lower frequencies and letting the high frequencies pass. The HP filter slope has a 12 dB/Octave roll-off.



The darker curve illustrates the roll-off curve of the Highpass Filter. The lighter curve in the middle represents the filter characteristic when the Resonance parameter is raised.

→ Notch

A notch filter (or band reject filter) could be described as the opposite of a bandpass filter. It cuts off frequencies in a narrow midrange band, letting the frequencies below and above through. On it's own, a notch filter doesn't really alter the timbre in any dramatic way, simply because most frequencies are let through. However, by combining a notch filter with a lowpass filter (using Filter 2 - see [page 109 in this chapter](#)), more musically useful filter characteristics can be created. Such a filter combination can produce soft timbres that still sound "clear". The effect is especially noticeable with low resonance (see [page 108](#)) settings.



The darker curve illustrates the roll-off curve of the Notch Filter. The lighter curve in the middle represents the filter characteristic when the Resonance parameter is raised.

Filter 1 Frequency

The Filter Frequency parameter (often referred to as "cutoff") determines which area of the frequency spectrum the filter will operate in. For a lowpass filter, the frequency parameter could be described as governing the "opening" and "closing" of the filter. If the Filter Freq is set to zero, none or only the very lowest frequencies are heard, if set to maximum, all frequencies in the waveform are heard. Gradually changing the Filter Frequency produces the classic synthesizer filter "sweep" sound.

! **Note that the Filter Frequency parameter is usually controlled by the Filter Envelope (see [page 110](#)) as well. Changing the Filter Frequency with the Freq slider may therefore not produce the expected result.**

Resonance

The filter resonance parameter is used to set the Filter characteristic, or quality. For lowpass filters, raising the filter Res value will emphasize the frequencies around the set filter frequency. This produces a generally thinner sound, but with a sharper, more pronounced filter frequency "sweep". The higher the filter Res value, the more resonant the sound becomes until it produces a whistling or ringing sound. If you set a high value for the Res parameter and then vary the filter frequency, this will produce a very distinct sweep, with the ringing sound being very evident at certain frequencies.

- For the highpass filter, the Res parameter operates just like for the lowpass filters.
- When you use the Bandpass or Notch filter, the Resonance setting adjusts the width of the band. When you raise the Resonance, the band where frequencies are let through (Bandpass), or cut (Notch) will become narrower. Generally, the Notch filter produces more musical results using low resonance settings.

Filter Keyboard Track (Kbd)

If Filter Keyboard Track is activated, the filter frequency will increase the further up on the keyboard you play. If a lowpass filter frequency is constant (a Kbd setting of "0") this can introduce a certain loss of "sparkle" in a sound the higher up the keyboard you play, because the harmonics in the sound are progressively being cut. By using a degree of Filter Keyboard Tracking, this can be compensated for.

Filter 2

A very useful and unusual feature of the Subtractor Synthesizer is the presence of an additional 12dB/Oct lowpass filter. Using two filters together can produce many interesting filter characteristics, that would be impossible to create using a single filter, for example formant effects.

The parameters are identical to Filter 1, except in that the filter type is fixed, and it does not have filter keyboard tracking.

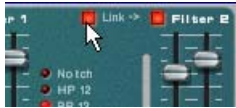
→ To activate Filter 2, click the button at the top of the Filter 2 section.

Filter 1 and Filter 2 are connected in series. This means that the output of Filter 1 is routed to Filter 2, but both filters function independently. For example, if Filter 1 was filtering out most of the frequencies, this would leave Filter 2 very little to “work with”. Similarly, if Filter 2 had a filter frequency setting of “0”, all frequencies would be filtered out regardless of the settings of Filter 1.



- ★ Try the “Singing Synth” patch (in the Monosynth category of the Factory Sound Bank) for an example of how dual filters can be used.

Filter Link



When Link (and Filter 2) is activated, the Filter 1 frequency controls the frequency *offset* of Filter 2. That is, if you have set different filter frequency values for Filter 1 and 2, changing the Filter 1 frequency will also change the frequency for Filter 2, but keeping the relative offset.

- ★ Try the “Fozzy Fonk” patch (in the Polysynth category of the Factory Sound Bank) for an example how linked filters can be used.

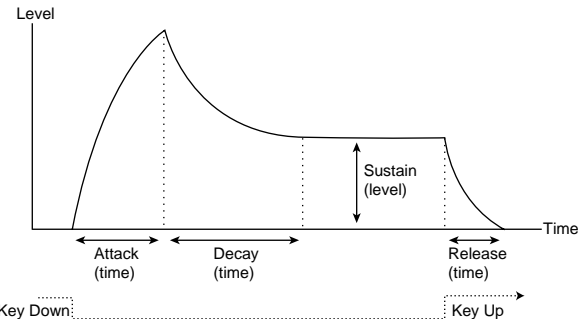
! Caution! If no filter modulation is used, and the filters are linked, pulling down the frequency of Filter 2 to zero will cause both filters to be set to the same frequency. If combined with high Res settings, this can produce very loud volume levels that cause distortion!

Envelopes - General

Envelope generators are used to control several important sound parameters in analog synthesizers, such as pitch, volume, filter frequency etc. Envelopes govern how these parameters should respond over time - from the moment a note is struck to the moment it is released.

Standard synthesizer envelope generators have four parameters; Attack, Decay, Sustain and Release (ADSR).

There are three envelope generators in the Subtractor, one for volume, one for the Filter 1 frequency, and one modulation envelope which has selectable modulation destinations.



The ADSR envelope parameters.

Attack

When you play a note on your keyboard, the envelope is triggered. This means it starts rising from zero to the maximum value. How long this should take, depends on the Attack setting. If the Attack is set to “0”, the maximum value is reached instantly. If this value is raised, it will take time before the maximum value is reached.

For example, if the Attack value is raised and the envelope is controlling the filter frequency, the filter frequency will gradually rise up to a point each time a key is pressed, like an “auto-wha” effect.

Decay

After the maximum value has been reached, the value starts to drop. How long this should take is governed by the Decay parameter.

If you wanted to emulate the volume envelope of a note played on a piano for example, the Attack should be set to "0" and the Decay parameter should be set to a medium value, so that the volume gradually decreases down to silence, even if you keep holding the key down. Should you want the decay to drop to some other value than zero, you use the Sustain parameter.

Sustain

The Sustain parameter determines the level the envelope should rest at, after the Decay. If you set Sustain to full level, the Decay setting is of no importance since the volume of the sound is never lowered.

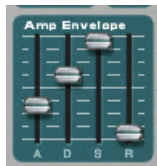
If you wanted to emulate the volume envelope of an organ, you theoretically only really need to use the Sustain parameter set to full level, as a basic organ volume envelope instantly goes to the maximum level (Attack "0") and stays there (Decay "0"), until the key is released and the sound instantly stops (Release "0").

But often a combination of Decay and Sustain is used to generate envelopes that rise up to the maximum value, then gradually decreases to finally land to rest on a level somewhere in-between zero and maximum. Note that Sustain represents a *level*, whereas the other envelope parameters represent times.

Release

Finally, we have the Release parameter. This works just like the Decay parameter, except it determines the time it takes for the value to fall back to zero *after* releasing the key.

Amplitude Envelope



The Amplitude Envelope is used to adjust how the volume of the sound should change from the time you press a key until the key is released. By setting up a volume envelope you sculpt the sound's basic shape with the four Amplitude Envelope parameters, Attack, Decay, Sustain and Release. This determines the basic character of the sound (soft, long, short etc.).

Filter Envelope



The Filter Envelope affects the Filter 1 Frequency parameter. By setting up a filter envelope you control the how the filter frequency should change over time with the four Filter Envelope parameters, Attack, Decay, Sustain and Release.

Filter Envelope Amount

This parameter determines to what degree the filter will be affected by the Filter Envelope. Raising this knob's value creates more drastic results. The Envelope Amount parameter and the set Filter Frequency are related. If the Filter Freq slider is set to around the middle, this means that the moment you press a key the filter is already halfway open. The set Filter Envelope will then open the filter further from this point. The Filter Envelope Amount setting affects *how much* further the filter will open.

Filter Envelope Invert



If this button is activated, the envelope will be inverted. For example, normally the Decay parameter lowers the filter frequency, but after activating Invert it will instead raise it, by the same amount.

Mod Envelope



The Mod Envelope allows you to select one of a number of parameters, or Destinations, to control with the envelope. By setting up a modulation envelope you control the how the selected Destination parameter should change over time with the four Mod Envelope parameters, Attack, Decay, Sustain and Release.

The available Mod Envelope Destinations are as follows:

Destination	Description
Osc 1	Selecting this makes the Mod Envelope control the pitch (frequency) of Osc 1.
Osc 2	Same as above, but for Osc 2.
Osc Mix	Selecting this makes the Mod Envelope control the oscillator Mix parameter. Both oscillators must be activated for this to have any effect.
FM	Selecting this makes the Mod Envelope control the FM Amount parameter. Both oscillators must be activated for this to have any effect.
Phase	Selecting this makes the Mod Envelope control the Phase Offset parameter for both Osc 1 and 2. Note that Phase Offset Modulation (Subtraction or Multiplication) must be activated for this to have any effect (see page 103).
Freq 2	Selecting this makes the Mod Envelope control the Frequency parameter for Filter 2.

LFO Section



LFO stands for Low Frequency Oscillator. LFOs are oscillators, just like Osc 1 & 2, in that they also generate a waveform and a frequency. However, there are two significant differences:

- LFOs only generate waveforms with low frequencies.
- The output of the two LFOs are never actually heard. Instead they are used for modulating various parameters.

The most typical application of an LFO is to modulate the pitch of a (sound generating) oscillator, to produce vibrato. Subtractor is equipped with two LFO's. The parameters and the possible modulation destinations vary somewhat between LFO 1 and LFO 2.

LFO 1 Parameters

Waveform

LFO 1 allows you to select different waveforms for modulating parameters. These are (from top to bottom):

Waveform	Description
Triangle	This is a smooth waveform, suitable for normal vibrato.
Inverted Sawtooth	This produces a "ramp up" cycle. If applied to an oscillator's frequency, the pitch would sweep up to a set point (governed by the Amount setting), after which the cycle immediately starts over.
Sawtooth	This produces a "ramp down" cycle, the same as above but inverted.
Square	This produces cycles that abruptly changes between two values, usable for trills etc.
Random	Produces random stepped modulation to the destination. On some vintage synths, this is called "sample & hold".
Soft Random	The same as above, but with smooth modulation.

Destination

The available LFO 1 Destinations are as follows:

Destination	Description
Osc 1&2	Selecting this makes LFO 1 control the pitch (frequency) of Osc 1 and Osc 2.
Osc 2	Same as above, but for Osc 2.
Filter Freq	Selecting this makes the LFO 1 control the filter frequency for Filter 1 (and Filter 2 if linked).
FM	Selecting this makes the LFO 1 control the FM Amount parameter. Both oscillators must be activated for this to have any effect.
Phase	Selecting this makes the LFO 1 control the Phase Offset parameter for both Osc 1 and 2. Note that Phase Offset Modulation (Subtraction or Multiplication) must be activated for this to have any effect (see page 103).
Osc Mix	Selecting this makes the LFO 1 control the oscillator Mix parameter.

Sync

By clicking this button you activate/deactivate LFO sync. The frequency of the LFO will then be synchronized to the song tempo, in one of 16 possible timedivisions. When sync is activated, the Rate knob (see below) is used for setting the desired timedivision.

Turn the knob and check the tooltip for an indication of the time division.



Rate

The Rate knob controls the LFO's frequency. Turn clockwise for a faster modulation rate.

Amount

This parameter determines to what degree the selected parameter destination will be affected by LFO 1. Raising this knob's value creates more drastic results.

LFO 2 Parameters

LFO 2 is polyphonic. This means that for every note you play, an *independent* LFO cycle is generated, whereas LFO 1 always modulates the destination parameter using the same "cycle". This can be used to produce subtle cross-modulation effects, with several LFO cycles that "beat" against each other. This also enables LFO 2 to produce modulation rates that vary across the keyboard (see the "Keyboard Tracking" parameter below).

Destination

The available LFO 2 Destinations are as follows:

Destination	Description
Osc 1&2	Selecting this makes LFO 2 modulate the pitch (frequency) of Osc 1 and Osc 2.
Phase	Selecting this makes the LFO 2 modulate the Phase Offset parameter for both Osc 1 and 2. Note that Phase Offset Modulation (Subtraction or Multiplication) must be activated for this to have any effect (see page 103).
Filter Freq 2	Selecting this makes the LFO 2 modulate the filter frequency for Filter 2.
Amp	Selecting this makes the LFO 2 modulate the overall volume., to create tremolo-effects.

LFO 2 Delay

This parameter is used to set a delay between when a note is played and when the LFO modulation "kicks in". For example, if Osc 1 & 2 is selected as the destination parameter and Delay was set to a moderate value, the sound would start out unmodulated, with the vibrato only setting in if you hold the note(s) long enough. Delayed LFO modulation can be very useful, especially if you are playing musical instrument-like sounds like violin or flute. Naturally it could also be used to control more extreme modulation effects and still retain the "playability" of the sound.

LFO 2 Keyboard Tracking

If LFO keyboard tracking is activated, the LFO rate will progressively increase the higher up on the keyboard you play. Raising this knob's value creates more drastic results.

🌀 If the LFO is set to modulate the phase offset, LFO keyboard tracking can produce good results. For example, synth string pads and other sounds that use PWM (see [page 103](#)) can benefit from this.

Rate

The Rate knob controls the LFO's frequency. Turn clockwise for a faster modulation rate.

Amount

This parameter determines to what degree the selected parameter destination will be affected by LFO 2. Raising this knob's value creates more drastic results.

Play Parameters

This section deals with two things: Parameters that are affected by how you play, and modulation that can be applied manually with standard MIDI keyboard controls.

These are:

- Velocity Control
- Pitch Bend and Modulation Wheel
- Legato
- Portamento
- Polyphony

Velocity Control



Velocity is used to control various parameters according to how hard or soft you play notes on your keyboard. A common application of velocity is to make sounds brighter and louder if you strike the key harder. Subtractor features very comprehensive velocity modulation capabilities. By using the knobs in this section, you can control how much the various parameters will be affected by velocity. The velocity sensitivity amount can be set to either positive or negative values, with the center position representing no velocity control.

The following parameters can be velocity controlled:

Destination	Description
Amp	This let's you velocity control the overall volume of the sound. If a positive value is set, the volume will increase the harder you strike a key. A negative value inverts this relationship, so that the volume decreases if you play harder, and increases if you play softer. If set to zero, the sound will play at a constant volume, regardless of how hard or soft you play.
FM	This sets velocity control for the FM Amount parameter. A positive value will increase the FM amount the harder you play. Negative values invert this relationship.
M. Env	This sets velocity control for the Mod Envelope Amount parameter. A positive value will increase the envelope amount the harder you play. Negative values invert this relationship.
Phase	This sets velocity control for the Phase Offset parameter. This applies to both Osc 1 & 2, but the relative offset values are retained. A positive value will increase the phase offset the harder you play. Negative values invert this relationship.
Freq 2	This sets velocity control for the Filter 2 Frequency parameter. A positive value will increase the filter frequency the harder you play. Negative values invert this relationship.
F. Env	This sets velocity control for the Filter Envelope Amount parameter. A positive value will increase the envelope amount the harder you play. Negative values invert this relationship.
F. Dec	This sets velocity control for the Filter Envelope Decay parameter. A positive value will increase the Decay time the harder you play. Negative values invert this relationship.
Osc Mix	This sets velocity control for the Osc Mix parameter. A positive value will increase the Osc 2 Mix amount the harder you play. Negative values invert this relationship.
A. Attack	This sets velocity control for the Amp Envelope Attack parameter. A positive value will increase the Attack time the harder you play. Negative values invert this relationship.

Pitch Bend and Modulation Wheels



The Pitch Bend wheel is used for "bending" notes, like bending the strings on a guitar. The Modulation wheel can be used to apply various modulation while you are playing. Virtually all MIDI keyboards have Pitch Bend and Modulation controls. Subtractor features not only the *settings* for how incoming MIDI Pitch Bend and Modulation wheel messages should affect the sound. Subtractor also has two functional wheels that could be used to apply real time modulation and pitch bend should you not have these controllers on your keyboard, or if you aren't using a keyboard at all. The Subtractor wheels mirror the movements of the MIDI keyboard controllers.

Pitch Bend Range

The Range parameter sets the amount of pitch bend when the wheel is turned fully up or down. The maximum range is "24" (=up/down 2 Octaves).

Modulation Wheel

The Modulation wheel can be set to simultaneously control a number of parameters. You can set positive or negative values, just like in the Velocity Control section. The following parameters can be affected by the modulation wheel:

Parameter	Description
F. Freq	This sets modulation wheel control of the Filter 1 Frequency parameter. A positive value will increase the frequency if the wheel is pushed forward. Negative values invert this relationship.
F. Res	This sets modulation wheel control of the Filter 1 Resonance parameter. A positive value will increase the resonance if the wheel is pushed forward. Negative values invert this relationship.
LFO 1	This sets modulation wheel control of the LFO 1 Amount parameter. A positive value will increase the Amount if the wheel is pushed forward. Negative values invert this relationship.
Phase	This sets modulation wheel control of the Phase Offset parameter for both Osc 1 and 2. Note that Phase Offset Modulation (Subtraction or Multiplication) must be activated for this to have any effect (see page 103).
FM	This sets modulation wheel control of the FM Amount parameter. A positive value will increase the FM amount if the wheel is pushed forward. Negative values invert this relationship. Both oscillators must be activated for this to have any effect.

Legato

Legato works best with monophonic sounds. Set Polyphony (see below) to 1 and try the following:

- ➔ **Hold down a key and then press another key without releasing the previous.**

Notice that the pitch changes, but the envelopes do not start over. That is, there will be no new “attack”.

- ➔ **If polyphony is set to more voices than 1, Legato will only be applied when all the assigned voices are “used up”.**

For example, if you had a polyphony setting of “4” and you held down a 4 note chord, the next note you played would be Legato. Note, however, that this Legato voice will “steal” one of the voices in the 4 note chord, as all the assigned voices were already used up!

Retrig

This is the “normal” setting for playing polyphonic patches. That is, when you press a key without releasing the previous, the envelopes are retriggered, like when you release all keys and then press a new one. In monophonic mode, Retrig has an additional function; if you press a key, hold it, press a new key and then release that, the first note is also retriggered.

Portamento (Time)

Portamento is when the pitch “glides” between the notes you play, instead of instantly changing the pitch. The Portamento knob is used to set how long it takes for the pitch to glide from one pitch to the next. If you don’t want any Portamento at all, set this knob to zero.

Setting Number of Voices - Polyphony



This determines the polyphony, i.e. the number of voices a Subtractor Patch can play simultaneously. This can be used to make a patch monophonic (⇒ a setting of “1”), or to extend the number of voices available for a patch. The maximum number of voices you can set a Subtractor Patch to use is 99. In the (unlikely) event you should need more voices, you can always create another Subtractor!

! Note that the Polyphony setting does not “hog” voices. For example, if you have a patch that has a polyphony setting of ten voices, but the part the patch plays only uses four voices, this won’t mean that you are “wasting” six voices. In other words, the polyphony setting is not something you need to consider much if you want to conserve CPU power - it is the number of voices *actually used* that counts.

About the Low Bandwidth button

This can be used to conserve CPU power. When activated, this function will remove some high frequency content from the sound of this particular device, but often this is not noticeable (this is especially true for bass sounds).

External Modulation



Subtractor can receive common MIDI controller messages, and route these to various parameters. The following MIDI messages can be received:

- Aftertouch (Channel Pressure)
- Expression Pedal
- Breath Control

If your MIDI keyboard is capable of sending Aftertouch messages, or if you have access to an Expression Pedal or a Breath controller, you can use these to modulate parameters. The “Ext. Mod” selector switch sets which of these message-types should be received.

These messages can then be assigned to control the following parameters:

Destination	Description
F. Freq	This sets External modulation control of the Filter 1 Frequency parameter. A positive value will increase the frequency with higher external modulation values. Negative values invert this relationship.
LFO 1	This sets External modulation control of the LFO 1 Amount parameter. A positive value will increase the LFO 1 amount with higher external modulation values. Negative values invert this relationship.
Amp	This let's you control the overall volume of the sound with external modulation. If a positive value is set, the volume will increase with higher external modulation values. A negative value inverts this relationship.
FM	This sets External modulation control of the FM Amount parameter. If a positive value is set, the FM amount will increase with higher external modulation values. A negative value inverts this relationship. Both oscillators must be activated for this to have any effect.

Connections



Flipping the Subtractor around reveals a plethora of connection possibilities, most of which are CV/Gate related. Using CV/Gate is described in the chapter “Routing Audio and CV”.

Audio Output

This is Subtractor’s main audio output. When you create a new Subtractor device, this is auto-routed to the first available channel on the audio mixer.

Sequencer Control

The Sequencer Control CV and Gate inputs allow you to play the Subtractor from another CV/Gate device (typically a Matrix or a Redrum). The signal to the CV input controls the note pitch, while the signal to the Gate input delivers note on/off along with velocity.

! For best results, you should use the Sequencer Control inputs with monophonic sounds.

Modulation Inputs

! Remember that CV connections will not be stored in the Subtractor patch, even if the connections are to/from the same Subtractor device!

These control voltage (CV) inputs (with associated voltage trim pots), can modulate various Subtractor parameters from other devices, or from the modulation outputs of the same Subtractor device. These inputs can control the following parameters:

- Oscillator Pitch (both Osc 1 & 2).
- Oscillator Phase Offset (both Osc 1 & 2).
- FM Amount
- Filter 1 Cutoff
- Filter 1 Res
- Filter 2 Cutoff
- Amp Level
- Mod Wheel

Modulation Outputs

The Modulation outputs can be used to voltage control other devices, or other parameters in the same Subtractor device. The Modulation Outputs are:

- Mod Envelope
- Filter Envelope
- LFO 1

Gate Inputs

These inputs can receive a CV signal to trigger the following envelopes. Note that connecting to these inputs will override the normal triggering of the envelopes. For example, if you connected an LFO output to the Gate Amp input, you would not trigger the amp envelope by playing notes, as this is now controlled by the LFO. In addition you would only hear the LFO triggering the envelope for the notes that you hold down. The following Gate Inputs can be selected:

- Amp Envelope
- Filter Envelope
- Mod Envelope



REASON

13

→ Malström Synthesizer

Introduction

The Malström is a polyphonic synthesizer with a great number of different routing possibilities. It is based on the concept of what we call "Grintable Synthesis" (see below), and is ideally suited for producing swirling, sharp, distorted, abstract special effect types of synthesizer sounds. In fact, you could go so far as to say that the Malström can produce sounds quite unlike anything you've ever heard from a synthesizer.

For a complete run-down of the principles behind it and thorough explanations of the controls, read on...

Features

The following are the basic features of the Malström:

- **Two Oscillators, based on Grintable Synthesis.**
See [page 121](#) for details.
- **Two Modulators, featuring tempo sync and one-shot options.**
See [page 123](#).
- **Two Filters and one Shaper.**
A number of different filter modes in combination with several routing options and a Waveshaper makes it possible to create truly astounding filter effects.
- **Three Envelope generators.**
There is one amplitude envelope for each oscillator and a common envelope for both filters. See [page 122](#) and [page 126](#) for details.
- **Polyphony of up to 16 voices.**
- **Velocity and Modulation control.**
See [page 131](#).
- **A number of CV/Gate Modulation possibilities.**
See [page 133](#).
- **A variety of Audio Input/Output options.**
You can for instance connect external audio sources for input to the Malström, and you can also control its output. See [page 132](#) for more details.

Theory of operation

There are a number of different synthesis methods for generating sound. There is e.g. Subtractive Synthesis (which is used in Reason's other synth - the Subtractor), FM Synthesis and Physical Modelling Synthesis to mention but a few.

To give you a clear understanding of the inner workings of the Malström, it might be in order with a brief explanation of what we call Grintable Synthesis.

What we refer to as Grintable Synthesis is actually a combination of two synthesis methods; Granular Synthesis and Wavetable Synthesis.

- In granular synthesis, sound is generated by a number of short, contiguous segments (grains) of sound, each typically between 5 to 100 milliseconds long. The sound is varied by changing the properties of each grain and/or the order in which they are spliced together. Grains can be produced either by a mathematical formula or by a sampled sound. This is a very dynamic synthesis method with a lot of variation possibilities, although somewhat hard to master and control.
- Wavetable synthesis on the other hand, is basically the playback of a sampled waveform. An oscillator in a wavetable synth plays back a single period of a waveform, and some wavetable synths also allow the possibility of sweeping through a set of periodic waveforms. This is a very straightforward synthesis method that is easily controlled, but somewhat limited in variation possibilities.

The Malström combines these two into a synthesis method that provides a very flexible way of synthesizing sounds with incredible flux and mutability.

It works like this:

- The oscillators in the Malström play back sampled sounds that are subject to some very complex processing and cut up into a number of grains. From here on, these sounds will be referred to as Grintables
- This results in a set of periodic waveforms (a grintable) that, when spliced together, play back the original sampled sound.
- This can then be treated just like a wavetable. I.e. It is possible to sweep through it. Move through it at any speed without affecting pitch. Play any section of it repeatedly. Use it to pick static waveforms. Jump between positions. Etc. etc.
- It is also possible to perform a number of other tricks, all of which are described further on in this chapter.

Loading and Saving Patches

Loading and saving patches is done in the same way as with any other Reason device. This is described in the chapter "Working with Patches" in the Getting Started book.



MALSTRÖM SYNTHESIZER

The Oscillator section



The two oscillators (osc:A and osc:B) of the Malström are the actual sound generators, and the rest of the controls are used for modulating and shaping the sound. The oscillators actually do two things; they play a graintable and generate the pitch:

- A graintable is several short, contiguous segments of audio (see above).
- Pitch is the frequency at which the segments are played back.

When creating a Malström patch, the fundamental first building block is usually to select a graintable for one or both of the oscillators.

- **To activate/deactivate an oscillator, click the On/Off button in the top left corner.**

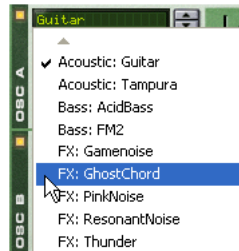
When an oscillator is activated, the button is lit.



An activated oscillator

- **To select a graintable, either use the spin controls or click directly in the display to bring up a pop-up menu with the available graintables.**

The graintables are sorted alphabetically into a number of descriptive categories, giving a hint as to the general character of the sound. Note that the categories are only visible in the pop-up menu, not in the display.



Setting oscillator frequency

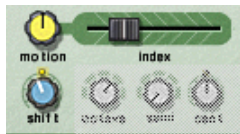
You can change the frequency - i.e. the tuning - of each oscillator by using the three knobs marked "Octave", "Semi" and "Cent".



- **The Octave knob changes the frequency in steps of one full octave (12 semitones).**
The range is -4 - 0 - +4 where 0 corresponds to middle "A" on your keyboard at 440 Hz.
- **The Semi knob changes the frequency in steps of one semitone.**
The range is 0 to +12 (one full octave up).
- **The Cent knob changes the frequency in steps of cents, which are 100ths of a semitone.**
The range is -50 - 0 - +50, i.e. down or up by up to half a semitone.

Controlling playback of the graintable

Each oscillator features three controls that determine how the loaded graintables are played back. These are: The "Index" slider, the "Motion" knob and the "Shift" knob.



- **The Index slider sets the playback starting point in the graintable.**
By dragging the slider, you set which index point in the graintable should be played first when the Malström receives a Note On message. Playback will then continue to the next index point according to the active graintable. With the slider all the way to the left, the first segment in the graintable is also the one that will be played back first.
- ! **Note that the Malström's Graintables are *not* all of the same length, and that the range for the Index slider (0-127) does *not* reflect the actual length of the graintables. I.e. regardless of whether a graintable contains 3 or 333 grains, the Index slider will always span the entire graintable even though the slider range says 0-127.**
- **The Motion knob controls how fast the Malström should move forward to play the next segment in the graintable, according to its motion pattern (see below).**
If the knob is kept in the middle position the speed of motion is the normal default. Turning the knob to the left slows it down and turning it to the right results in higher speed. If the knob is set all the way to the left, there will be no motion at all, which means that the initial segment, as set with the Index slider, will play over and over as a static waveform.
- **The Shift knob changes the timbre of the sound (the formant spectrum).**
What it actually does is change the pitch of a segment up or down by re-sampling. However, since the pitch you *hear* is independent of the actual pitch of the graintable (see above), pitch-shifting a segment instead means that more or less of the segment waveform will be played back, resulting in a change of harmonic content and timbre.

About motion patterns

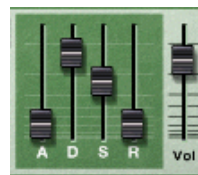
Each graintable has a predefined motion pattern and a default motion speed.

When a graintable is looped (i.e. if the Motion knob is *not* set all the way to the left), it follows one of two possible motion patterns:

- **Forward**
This motion pattern plays the graintable from the beginning to the end, and then repeats it.
- **Forward - Backward**
This motion pattern plays the graintable from the beginning to the end, then from the end to the beginning and then repeats it.

The motion speed can be changed with the Motion knob, as described above, but it is *not* possible to alter the motion pattern of a graintable.

The amplitude envelopes



Each oscillator features a standard ADSR (Attack, Decay, Sustain, Release) envelope generator, and a Level control. These are used for controlling the volume of the oscillator. One thing that makes the Malström different from many other synths though, is the fact that the amplitude envelopes are placed *before* the filter and routing sections in the signal path.

The amplitude envelopes control how the volume of a sound should change from the moment you strike a key on your keyboard to the moment that you release it again.

Vol

The Volume knobs set the volume level out from each oscillator.

- ! **For an overall description of the general envelope parameters (Attack, Decay, Sustain, Release), please refer to the Subtractor chapter.**

The Modulator section



The Malström features two Modulators (mod:A and mod:B) These are in fact another type of oscillators, called LFOs (Low Frequency Oscillators). They each generate a waveform and a frequency, much like osc:A and osc:B. However, there are a couple of important differences:

- Mod:A and mod:B do not generate sound. They are instead used for modulating various parameters to change the character of the sound.
- They only generate waveforms of low frequency.

Furthermore, both modulators are tempo syncable and possible to use in one shot mode, in which case they will actually work like envelopes.

Modulator parameters

The two Modulators have a few controls in common, but there are also some differences. Both the common parameters and the ones that are unique for each Modulator (the destinations) are described below.

→ **To activate/deactivate a Modulator, click the On/Off button in the top left corner.**

When a Modulator is activated, the button is lit.



An activated Modulator

Curve

This lets you select a waveform for modulating parameters. Use the spin controls to the right of the display to cycle through the available waveforms. Some of these waveforms are especially suited for use with the Modulator in one shot mode (see below).

Rate

This knob controls the frequency of the Modulator. For a faster modulation rate, turn the knob to the right.

The Rate knob is also used for setting the time division when synchronizing the Modulator to the song tempo (see below).

One Shot

To put the Modulator into one shot mode, click this button so that it is lit.

Normally, the Modulators will repeat the selected waveforms over and over again, at the set rate. However, when one shot mode is activated and you play a note, the Modulator will play the selected waveform only once (at the set rate) and then stop. In other words, it will effectively be turned into an envelope generator!

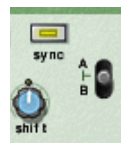
Note that even though all waveforms can be used with interesting results, some waveforms are explicitly well suited for use in one shot mode. For example, try using the waveform with just one long, gently sloping curve.

Sync

Clicking this button so that it is lit synchronizes the Modulator to the song tempo, in one of 16 possible time divisions.

! When sync is activated, the Rate knob is used for selecting the desired timedivision. Turn the Rate knob and observe the tool tip for an indication of the timedivision.

A/B selector



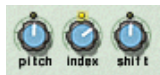
This switch is used for deciding which oscillator and/or filter the Modulator should modulate - A, B or both. With the switch in the middle position, both A and B will be modulated.

Destinations

The following knobs are used for determining what each of the two modulators should modulate.

- Note that these knobs are bi-polar, which means that if a knob is in the middle position, no modulation is applied. If you turn a knob either to the left or to the right, an increasing amount of modulation is applied to the parameter. The difference is that if you turn a knob to the left, the waveform of the modulator is inverted.

Mod:A



Mod:A can modulate the following parameters of either oscillator:

- **Pitch**
Use this if you want Mod:A to offset the pitch of osc:A, osc:B, or both (see [page 121](#)).
- **Index**
Use this if you want Mod:A to offset the index start position of osc:A, osc:B, or both (see [page 122](#)).
- **Shift**
Use this if you want Mod:A to affect the harmonic content of osc:A, osc:B, or both (see [page 122](#)).

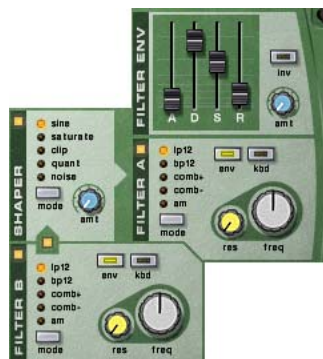
Mod:B



Mod:B can modulate the following parameters of either oscillator:

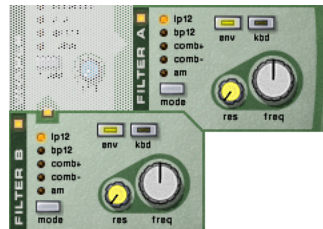
- **Motion**
Use this if you want Mod:B to affect the motion speed of osc:A, osc:B, or both (see [page 122](#)).
- **Level**
Use this if you want Mod:B to change the output level of osc:A, osc:B, or both (see [page 122](#)).
- **Filter**
Use this if you want Mod:B to offset the cutoff frequency of filter:A, filter:B, or both (see [page 126](#)).
- **Mod:A**
Use this if you want Mod:B to change the total amount of modulation from Mod:A.

The Filter section



The filter section lets you further shape the overall character of the sound. Contained herein are two multimode filters, a filter envelope and a waveshaper.

The Filters



Both filter:A and filter:B have the exact same parameters, all of which are described below

- **To activate/deactivate a filter, click the On/Off button in the top left corner.**

When a filter is activated, the button is lit.



An activated filter

Filter types

To select a filter type, either click the Mode button in the bottom left corner or click directly on the desired filter name so that it lights up in yellow:

- **LP 12 (12 dB lowpass)**

Lowpass filters let low frequencies through and cut off high frequencies. This filter type has a roll-off curve of 12dB/Octave.

- **BP 12 (12 dB bandpass)**

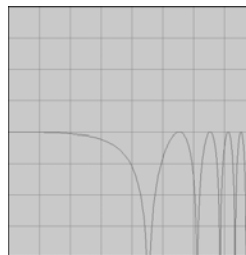
Bandpass filters cut both high and low frequencies, leaving the frequency band in between unaffected. Each slope in this filter type has a 12 dB/Octave roll-off.

- **Comb + & Comb -**

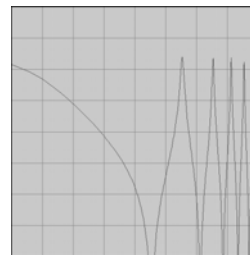
Comb filters are basically delays with very short delay times with adjustable feedback (in Reason controlled with the Resonance knob). A comb filter causes resonating peaks at certain frequencies.

The difference between “+” and “-” is in the position of the peaks, in the spectrum. The main audible difference is that the “-”-version causes a bass cut.

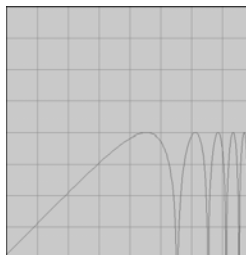
The Resonance parameter in both cases controls the shape and size of the peaks.



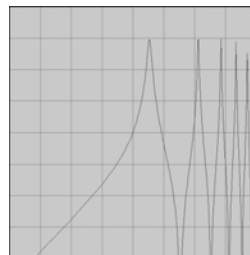
Comb + Low Resonance



Comb + High Resonance



Comb - Low Resonance



Comb - High resonance

- **AM**

AM (Amplitude Modulation) is often referred to as Ring Modulation. A Ring Modulator works by multiplying two signals together. In the case of the Malström, the filter produces a sine wave which is multiplied with the signal from osc:A or osc:B. Resonance controls the mix between the clean and modulated signals. The Ring Modulated output will then contain added frequencies which are generated by the sum of, and the difference between the two signals. This can be used for creating complex, non-harmonic sounds.

Filter controls

Each filter contains the following four controls:

→ Kbd (keyboard tracking)

By clicking this button so that it is lit, you activate keyboard tracking. If keyboard tracking is activated, the frequency of the filter will change according to the notes you play on your keyboard. That is, if you play notes higher up on the keyboard, the filter frequency will increase and vice versa. If keyboard tracking is deactivated, the filter frequency will remain at a fixed value regardless of where on the keyboard you play.

→ Env (envelope)

If you click on this button so that it is lit, the cutoff frequency (see below) will be modulated by the filter envelope. If you leave this deactivated, the Filter Envelope will have no effect.

→ Freq (frequency)

The function of this parameter depends on which filter type you have selected:

With all filter types except AM, it is used for setting the cutoff frequency of the filter. In the case of the lowpass filter for example, the cutoff frequency determines the limit above which high frequencies will be cut off. Frequencies below the cutoff frequency will be allowed to pass through. The farther to the right you turn the knob, the higher the cutoff frequency will be.

If you have selected AM as filter type, this will instead control the frequency of the signal generated by the filter. The same control range applies though; the farther to the right you turn the knob the higher the frequency will be.

→ Res (resonance)

Again, the function of this parameter depends upon which filter type is selected:

If the selected filter is any other type than AM, it sets the filter characteristic, or quality. For the lowpass filter for example, raising the filter Res value will emphasize the frequencies around the set filter frequency. This generally produces a thinner sound, but with a sharper, more pronounced filter frequency “sweep”. The higher the filter Res value, the more resonant the sound becomes until it produces a whistling or ringing sound. If you set a high value for the Res parameter and then vary the filter frequency, this will produce a very distinct sweep, with the ringing sound being very evident at certain frequencies.

In the case of the AM filter type though, this control instead regulates the balance between the original signal and the signal resulting from amplitude modulation. The farther to the right you turn the knob, the more dominant the AM signal will be.

The Filter Envelope



This is a standard ADSR envelope with two additional controls; inv and amt.

The filter envelope is common for both filter:A and filter:B, and controls how the filter frequency should change over time.

Inv (inverse)

This button toggles inversion of the envelope on and off. The Decay segment of the envelope will for instance normally lower the frequency, but if the envelope is inverted it will instead raise the frequency.

Amt (amount)

This controls to which extent the filter envelope affects the filters, or rather - the set filter cutoff frequencies. For example; if the cutoff frequency is set to a certain value, the filter will already be opened by this amount when you hit a key on your keyboard. The amount setting then controls how much more the filter will open from that point. Turn the knob to the right to increase the value.

! **For an overall description of the general envelope parameters (Attack, Decay, Sustain, Release), please refer to the Subtractor chapter.**

The Shaper



Before filter: A is an optional waveshaper. Waveshaping is a synthesis method for transforming sounds by altering the waveform shape, thereby creating a complex, rich sound. Or, if that's more to your taste, truncating and distorting the sound to lo-fi heaven!

A guitar distortion box could be viewed as a type of waveshaper for example. An unamplified electric guitar produces a sound with fairly pure harmonic content, which is then amplified and transformed by the distortion box.

- **To activate/deactivate the Shaper, click the On/Off button in the top left corner.**

When the Shaper is activated, the button is lit.



The Shaper activated

Mode

You can select one of five different modes for shaping the sound, each with its own characteristics.

To select a mode, either click the Mode button in the bottom left corner or click directly on the desired mode name so that it lights up in yellow.

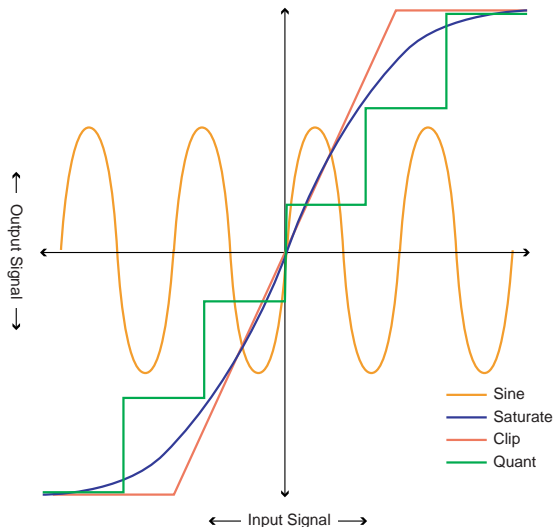
- **Sine**
This produces a round, smooth sound.
- **Saturate**
This gives a lush, rich character to the sound.
- **Clip**
This introduces clipping - digital distortion - to the signal.

→ Quant

This lets you truncate the signal by bit-reduction, thus making it possible to achieve that noisy, characteristic 8 bit sound for example.

→ Noise

This is actually not strictly a shaper function. Instead it multiplies the sound with noise.



Amt (amount)

This controls the amount of shaping applied. By turning the knob to the right you increase the effect.

Routing

The Malström puts you in total control of how the signal should be routed from the oscillators, through the filters and on to the outputs. Below is first a general description of the routing options, followed by examples of how to route the signal in order to achieve a certain result.

→ **Click on a button so that it is lit, to route the signal correspondingly.**
See below for descriptions.

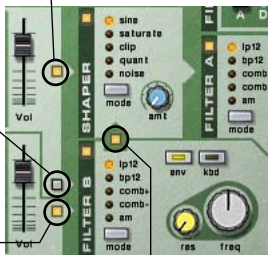
If this button is lit, the signal from osc:A is routed to filter:A via the shaper. If neither this nor the other routing button from osc:A (to filter:B) is lit, the signal will go straight to the outputs.

If this button is lit, the signal from osc:A is routed to filter:B. If neither this nor the other routing button from osc:A (to filter:A/shaper) is lit, the signal from osc:A will go straight to the outputs.

If this button is lit, the signal from osc:B is routed to filter:B. If this is not lit, the signal from osc:B will go straight to the outputs.

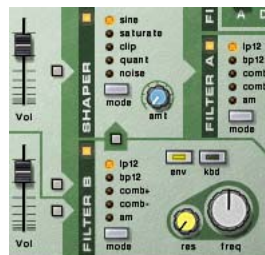
If this button is lit, the signal from filter:B is routed to filter:A via the shaper. The signal from filter:B can originate from either osc:A, osc:B or both. If this is not lit, the signal from filter:B will go straight to the outputs.

! **Note that the result depends both on the routing buttons and on whether the filters and shaper are activated or not!**



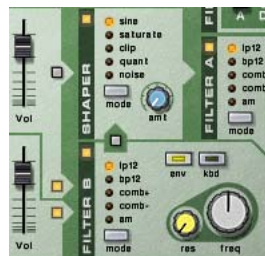
Routing examples

One or both oscillators without filters

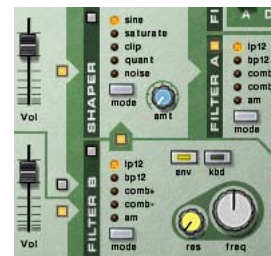


With this configuration, the signals from the oscillators will bypass the filters and go directly to the respective output. Using both oscillators allows you to use the Spread parameter to create a true stereo sound.

One or both oscillators to one filter only



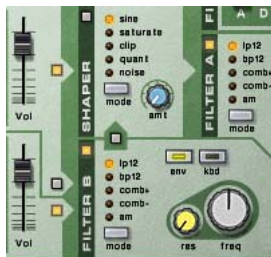
Both oscillators routed to filter:B only.



Both oscillators routed to filter:A only.

With these configurations, the signal from osc:A and/or osc:B will go to either filter:A or filter:B and then to the outputs. This is essentially a mono configuration and hence Spread should probably be set to "0".

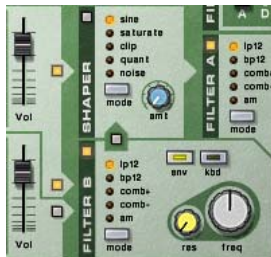
Both oscillators with one filter each



With this configuration, the signals from osc:A and osc:B will go to filter:A and filter:B respectively, and then to the outputs.

Again, this configuration allows you to work in true stereo.

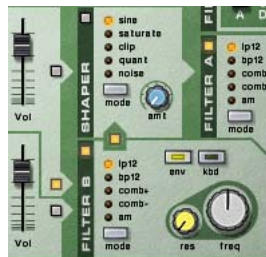
One oscillator with both filters in parallel



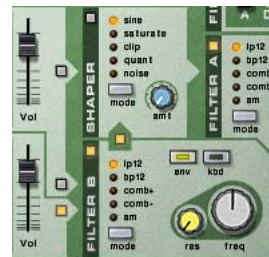
With this configuration, the signal from osc:A will go to both filter:A and filter:B, with the filters in parallel.

! This configuration is only possible with osc:A. Osc:B can be routed to both filters as well, but only in series (see below).

One or both oscillators with both filters in series



Osc:A routed through both filters in series.

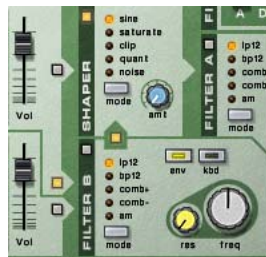


Osc:B routed through both filters in series.

With these configurations, the signal from osc:A and/or osc:B will go to both filter:A and filter:B, with the filters in series (one after the other).

Adding the shaper

The signal from one or both oscillators can also be routed to the shaper. The signal will then pass through the shaper to the outputs, with or without also passing through the filters.



In the left figure, the signal from osc:A is routed to the shaper and then directly to the outputs. In the right figure, the signal from osc:B is routed to filter:B, then to the shaper and then to filter:A.

The output controls



These two parameters control the output from the Malström in the following way:

Volume

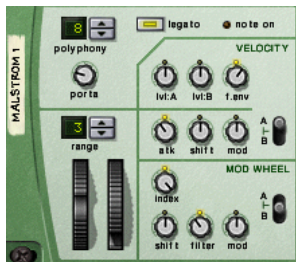
This knob controls the master volume out from the Malström.

Spread

This controls the stereo pan-width of the outputs from Osc:A/B and Filter:A/B respectively. The farther to the right you turn the knob, the wider the stereo image will be. In other words, the signals will be panned further apart to the left and right.

! If you are only using one output (A or B), it is strongly recommended that you set Spread to “0”.

The play controls



To the far left on the Malström's “control panel” are various parameters that are affected by how you play, and lets you apply modulation by MIDI controls. The following is a description of these controls.

Polyphony - setting the number of voices



This lets you set the polyphony for the Malström. Polyphony is the number of voices it can play simultaneously. The maximum number is 16 and the minimum is 1, in which case the Malström will be monophonic.

! The number of voices you can play depends of course on the capacity of your computer. Even though the maximum number is 16 it doesn't necessarily mean that your system is capable of using that many voices. Also note that voices do not consume CPU capacity unless they are really “used”. That is, if you are using a patch that plays two voices but have polyphony set to four, the two “unused” voices do not consume any of your system resources.

Porta (portamento)



This is used for controlling portamento. This is a parameter that makes the pitch glide between the notes you play, rather than changing the pitch instantly as soon as you hit a key on your keyboard. By turning this knob you set how long it should take for the pitch to glide from one note to the next as you play them.

With the knob turned all the way to the left, portamento is disabled.

Legato



By clicking this button you activate/deactivate Legato. Legato in Malström is unique in that it allows you to control whether the sound is monophonic or polyphonic by using your playing style:

- **If you play legato (hold down a key and then press another key without releasing the previous), the sound is monophonic.**
Also note that the pitch changes, but the envelopes do not start over. That is, there will be no new “attack”.
- **If you play non-legato (separated notes), with polyphony set to more voices than 1, each note will decay separately (polyphonic).**
This will be most apparent with longer release times.

The Pitch Bend and Modulation wheels



- The Pitch Bend wheel is used for bending the pitch of notes, much like bending the strings on a guitar or other string instrument.
- The Modulation wheel can be used for applying modulation while you are playing.

Virtually all MIDI keyboards have Pitch Bend and Modulation controls. The Malström does not only feature the settings for how incoming MIDI Pitch Bend and Modulation wheel messages should affect the sound, but also two functional wheels that can be used for applying real time modulation and pitch bend if you don't have these controllers on your keyboard, or if you aren't using a keyboard at all. The wheels on the Malström also mirror the movements of the wheels on your MIDI keyboard.

Pitch Bend Range

The Range parameter sets the maximum amount of pitch bend, i.e. how much it is possible to change the pitch by turning the wheel fully up or down. The maximum range is 24 semitones (2 Octaves). You change the value by clicking the spin controls to the right of the display.

The Velocity controls



Velocity is used for controlling various parameters according to how hard or soft you play notes on your keyboard. A typical use of velocity control is to make sounds brighter and louder if you strike a key harder. By using the knobs in this section, you can control how much the various parameters will be affected by velocity.

! All of the velocity control knobs are bi-polar, which means that the amount can be set to either positive or negative values, while keeping the knobs in the center position means that no velocity control is applied.

The following parameters can be velocity controlled:

- **Lvl:A**
This lets you velocity control the output level of osc:A.
 - **Lvl:B**
This lets you velocity control the output level of osc:B.
 - **F.env**
This sets velocity control for the Filter Envelope Amount parameter. Positive values will increase the envelope amount the harder you play, and negative values will decrease the amount.
 - **Atk (attack)**
This sets velocity control for the Amp Envelope Attack parameter of osc:A and/or osc:B. Positive values will increase the Attack time the harder you play, and negative values will decrease it.
 - **Shift**
This lets you velocity control the Shift parameter of osc:A and/or osc:B.
 - **Mod**
This lets you velocity control all modulation amounts of mod:A and/or mod:B.
- ! Note that you can set the last three parameters (Atk, Shift and Mod) to be velocity controlled for either or both of oscillator/modulator A and B. This is done with the A/B selector switch.**

The Modulation wheel controls



The Modulation wheel can be set to control a number of parameters. You can set positive or negative values, just like in the Velocity Control section (see above).

The following parameters can be affected by the modulation wheel:

→ Index

This sets modulation wheel control of the currently active graintable's index (see [page 122](#)) for osc:A and/or osc:B. Positive values will move the index position forwards if the modulation wheel is pushed forward. Negative values will move it backwards.

→ Shift

This sets modulation wheel control of the Shift parameter of osc:A and/or osc:B (see [page 122](#)).

→ Filter

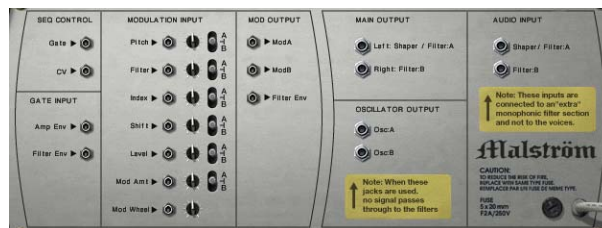
This sets modulation wheel control of the Filter Frequency parameter (see [page 126](#)). Positive values will raise the frequency if the wheel is pushed forward and negative values will lower the frequency.

→ Mod

This sets modulation wheel control of the total amount of modulation from mod:A and/or mod:B. Positive values will increase the settings if the wheel is pushed forward and negative values will decrease the settings.

! You can set whether these parameters on either or both oscillator/modulator/filter A and B will be affected by the modulation wheel. This is done with the A/B selector switch.

Connections



Flipping the Malström around reveals a wide array of connection possibilities. Most of these are CV/Gate related. Using CV/Gate is described in the chapter "Routing Audio and CV".

Audio Output

These are the Malström's audio outputs. When you create a new Malström device, they are auto-routed to the first available channel on the audio mixer:

→ Shaper/Filter:A (left) & Filter:B (right)

These are the main stereo outputs. Each of the two filters are connected to a separate output, and by connecting both, you can have stereo output. Whether the output really will be in stereo however, is determined by the routing and the Spread parameter. See [page 128](#) for details about this.

→ Osc:A & osc:B

These make it possible to output the sound directly after the Amp Envelope of each oscillator, bypassing the filter section. Connecting one or both of these to a channel on the audio mixer will break the Malström's internal signal chain. That is, it is not possible to process the sound by using the filters and the shaper of the Malström, the sound instead goes directly to the mixer.

★ **Note also that you can connect the outputs Osc:A & Osc:B to the Audio Inputs on the Malström for some interesting effects - see [page 133](#).**

Audio Input

- Shaper/Filter:A
- Filter:B

These inputs let you connect either other audio sources, or the Malström's own internal signal directly to the filters and the shaper - see [page 133](#).

Sequencer Control

The Sequencer Control CV and Gate inputs allow you to play the Malström from another CV/Gate device (typically a Matrix or a Redrum). The signal to the CV input controls the note pitch, while the signal to the Gate input delivers note on/off along with velocity.

! **For best results, you should use the Sequencer Control inputs with monophonic sounds.**

Gate Input

These inputs can receive a CV signal to trigger the following envelopes:

- Amp Envelope
- Filter Envelope

! **Note that connecting to these inputs will override the normal triggering of the envelopes. For example, if you connected a Modulation output to the Gate Amp input, you would not trigger the amp envelope by playing notes, as this is now controlled by the Modulator. In addition you would only hear the Modulator triggering the envelope for the notes that you hold down.**

Modulation Input

These control voltage (CV) inputs (with associated voltage trim pots and A/B selector switches), can modulate various Malström parameters from other devices, or from the modulation outputs of the same Malström device. These inputs can control the following parameters:

- Oscillator Pitch
- Filter Frequency
- Oscillator Index offset
- Oscillator Shift
- Amp Level
- Mod Amount
- Mod Wheel

Modulation Output

The Modulation outputs can be used to voltage control other devices, or other parameters in the same Malström device.

The Modulation Outputs are:

- Mod:A
- Mod:B
- Filter Envelope

Routing external audio to the filters



The audio inputs on the back of the Malström allows you to connect any audio signal to the filters and Shaper.

To use this feature, it's important to understand the following background:

Normally the Malström behaves like any regular polyphonic synthesizer, in that each voice has its own filter. The filter settings are the same, but each filter envelope is triggered individually when you play a note.

However, when you connect a signal to the audio inputs, it is routed to an "extra" filter. The envelope for this filter is triggered each time *any* of the other filter envelopes is triggered. In other words, the "extra" filter envelope is triggered each time you play a note on the Malström.

There are two different uses for the audio inputs:

Connecting an external signal source

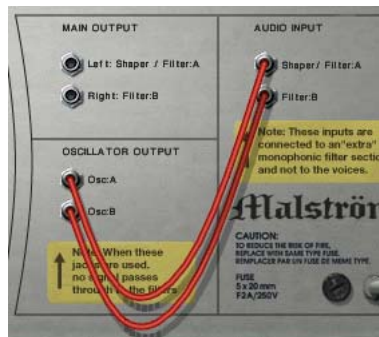
Connecting an audio signal from another device in the rack to the audio input allows you to process the signal through the filters and/or Shaper of the Malström. The processed signal will then be mixed with the Malström's "own" voices (if activated) and sent to the outputs.

The result depends on the following:

- To which jack you connect the signal.
- Whether the filters and/or Shaper are activated on the front panel.
- The routing button for filter:B.
If this is activated and you connect a signal to the Filter:B input, the signal will be processed in filter:B and then sent to the Shaper and filter:A (just as when routing Malströms own oscillators on the front panel).

Note again that the filter envelope is triggered by all voices. To make use of the filter envelope, you either need to play the Malström or use gate signals to trigger it or the filter envelope, separately.

Connecting the signal from the Malström itself



If you connect one or both oscillator outputs to the audio input(s), the internal signal path from the oscillators to the filters is broken. In other words, no signals will pass internally from the oscillators to the filters, and the three routing buttons for the oscillators are ignored.

This may seem pointless at first, but there are several uses for this:

- **When you play the Malström in this mode, the filter envelope will be triggered for each note you play, affecting all sounding notes.**

This is due to the monophonic “extra” filter described above. On older synthesizers, this feature is called “Multiple triggering”.

- **Since all notes you play are mixed before being sent into the filter, the result of using the Shaper will be totally different (if you play more than one note at a time).**

This is similar to playing a guitar chord through a distortion effect, for example.

- **You can patch in external effects between the oscillators and the filters.**

Just connect an oscillator output to the input of the effect device, and the effect output to the Malström's audio input.

- ★ **You can use combinations of connections and routing. You could for instance connect an external audio signal to one of the inputs, one of the Malström's oscillators to the other input and then use the routing options on the front panel for the other oscillator. All of these signals will then be mixed and sent to the Malström's main outputs.**



REASON

14

→ NN-19 Sampler

Introduction

A sampler could be described as a device capable of recording and reproducing audio material, like a tape recorder. Unlike a tape or hard disk based recorder, samplers allows you to “play” the recorded sound via MIDI, using a keyboard for example. This way, any reproducible sound can be integrated into the MIDI environment, and be controlled from sequencers etc. like synthesizers.

The NN-19 is a sample *player*, capable of reproducing, but not recording or editing sound files.

The program comes with more than a hundred ready-made sample patches, covering all kinds of instrument types. In addition to this there are plenty of single samples that can be used for creating your own patches.

If you want to record or edit your own samples, there are plenty of relatively inexpensive (and even free) audio editing software for both the Windows and the Mac OS platforms, that will allow you to both record audio (via your computers or audio cards audio inputs), and to edit the resulting audio file. Virtually every product that is capable of this, can create sound files which can be loaded directly into the NN-19.

Also, there are thousands of high quality sample CD:s available, covering every conceivable musical style or direction ranging from professionally recorded orchestral samples to esoteric electronic noises.

General Sampling Principles

Background

Before a sound can be used by a sampler, it must be converted to a digital signal. Hardware samplers provide audio inputs that can convert the analog signal to digital, by the use of an “A/D Converter” (analog to digital). This “samples” the signal at very short time intervals and converts it to a digital representation of the analog signal’s waveform. The sample rate and the bit depth of this conversion determines the resulting sound quality. Finally the signal is passed through a digital to analog converter (D/A) which reconstructs the digital signal back to analog, which can be played back.

Multisampling vs. Single Samples

Most of the included NN-19 patches are made up of a collection of several samples. This is because a single sampled sound only sounds natural within a fairly narrow frequency range. If a single sample is loaded into an empty NN-19, the sample will be playable across the whole keyboard. The pitch (frequency) of the original sample (called root-key) will be automatically placed on the middle C key (C3).

Note that this has nothing to do with the actual pitch the sample itself produces! It may not even have a pitch as such, it could be the sound of someone talking for example.

If you play any single sample about two octaves above or below its root key, it will most likely sound very “unnatural”. In the case of it actually being a sample of someone talking, playing two octaves up will make the talking voice sample sound squeaky, short and most likely unintelligible. Two octaves down the voice will sound something like a drawn-out gargle.

Thus, the range that most samples can be transposed without sounding unnatural is limited. To make a sampled piano, for example, sound good across the whole keyboard, you need to first have made many samples at close intervals across the keyboard, and then define an upper and lower range for each sample, called a *Key Zone*. All the keyzones in the piano sample patch then make up a *Key Map*.

How to create key zones is described on [page 138](#).

To sample real instruments accurately requires a lot of hard work. Firstly, you need the original instrument, which should be in perfect working order. For acoustic instruments you need a couple of good microphones, a mixer or other device with high quality microphone preamps, and a room with good acoustics. You need to be meticulous when recording the different samples, so that levels are smooth and even across the range etc.

Fortunately Reason provides a wide range of high quality multisampled instruments, so much of this hard work has already been done for you.

In our experience, most people don’t use samplers only for playing sampled versions of “real” instruments. Very often, single “stand alone” or single samples are used. Maybe you wish to use *different* sounds for every key zone. Or you could have complete chorus and verse vocals plus variations assigned to several “one note” key zones. Or use samples of different chords that play rhythmic figures to the same tempo, and use these to build song structures etc. The possibilities are endless. When you use samples in this way, the keys on your keyboard that play the samples do not necessarily correspond to pitch at all, the keys are simply used to *trigger* the samples.

About Audio File Formats

The NN-19 can read audio files in the following formats:

- Wave (.wav)
- AIFF (.aif)
- SoundFonts (.sf2)
- REX file slices (.rex2, .rex, .rcy)
- Any sample rate and practically any bit depth.

! If you want the files to play back with their original bit depth - if higher than 16-bits - make sure to activate “Use High Resolution Samples” on the General page in the Preferences dialog. Otherwise, samples will be played back as 16-bit files in NN-19 regardless of their original bit depth. See the Getting Started book for more details.

Wave and AIFF are the standard audio file formats for the PC and Mac platforms, respectively. Any audio or sample editor, regardless of platform, can read and create audio files in at least one of these formats.

SoundFonts are an open standard for wavetable synthesized audio, developed by E-mu systems and Creative Technologies.

REX files are music loops created in the ReCycle program (see below). The NN-19 lets you either load REX files as patches or separate slices from REX files as individual samples.

About the Sample Patch format

Reason's Sample Patch format (.smp), is based on either Wave or AIFF files, but includes all the NN-19 associated parameter settings as well.

→ **The audio files may be stereo or mono. Stereo audio files are shown with a “S” symbol beside its name in the display.**

Loading a Sample Patch

When you create a new NN-19 device, it is empty. That is, the “Init patch” in the NN-19 does not contain any samples. For NN-19 to produce sound, you need to load either a sample patch, or a sample.

A patch contains “everything”. All the samples, assigned key zones, and associated panel settings will be loaded. Loading a sample patch is done using the patch browser, just like in all other devices that use Patches.

Open the folder that contains the NN-19 patch you wish to load, select it and click open.

Loading REX Files as Patches

REX Files are files created in the ReCycle program. This is an application created by Propellerhead Software, used for slicing up music loops and enabling them to be played back in any tempo. In Reason, REX files are primarily used in the Dr. Rex loop player, but they can be used in the NN-19 as well. Possible extensions are “.rx2”, “.rcy” and “.rex”.

When loading a REX file, each slice in the file is assigned to one key, chromatically. All parameters are set to default settings.

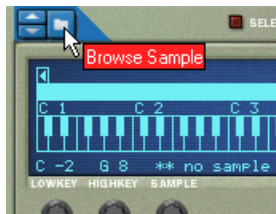
When using REX files in the DR. Rex loop player, it is possible to make a track play the slices in order to recreate the original loop. To do the same in the NN-19 requires a few extra steps.

1. **Use the patch browser to load the REX file into an NN-19 sampler.**
2. **Create a Dr. Rex loop player and load the same REX file in to this device.**
3. **Use the To Track feature on the Dr. Rex to create playback data (a group) on the track assigned to the Dr. Rex.**
4. **Move that group to the track that plays the NN-19 and play it back from there.**
5. **Delete the Dr. Rex loop player.**

About Key Zones and Samples

Loading a Sample into an empty NN-19

1. Create a new sampler device.
2. Click on the sample browser button.
This is located above the keyboard display to the left.



- ⚙️ When you browse samples, you can preview them before loading using the browser Play button. If you select the Preview “Autoplay” function, the samples play back once automatically when selected.

3. Use the browser to select a sample and open it.

When you load the first sample into an empty NN-19, this will be assigned a key zone that spans the entire range of the keyboard, and the default Init Patch settings will be used.

Below the keyboard, the range, sample name, root key, tuning, level and loop status of the current key zone is displayed, each with a corresponding knob.



The light blue strip above the keyboard indicates the currently selected key zone, which is in this case the full range of the keyboard.

The inverted note on the keyboard indicates the “root key” of the sample. All samples contain a root key, tuning and level setting. If NN-19 is empty, a sample will have its root key placed on the middle “C” (C3) key.

4. If desired, click on the keyboard to change the root key.

! You can audition a loaded sample patch or sample by holding down [Option] (Mac)/[Alt] (Windows) and clicking on a key in the Keyboard display. The mouse will take on the shape of a speaker symbol to indicate this.

Loading SoundFont samples

The SoundFont format was developed by E-mu systems in collaboration with Creative Technologies. It is a standardized data format containing wavetable synthesized audio and information on how it should be played back in wavetable synthesizers - typically on audio cards. The SoundFont format is an open standard so there is a vast amount of SoundFont banks and SoundFont compatible banks developed by third parties.

The samples in a SoundFont are stored hierarchically in different categories: User Samples, Instruments, Presets etc. The NN-19 allows you to browse for and load single SoundFont samples, but *not* entire soundfonts.

1. Use the sample browser to select a SoundFont file (.sf2) and open it.

The browser opens the SoundFont and displays the folders within it.

2. Select the folder “Samples” and open it.

This folder contains a number of samples which can be loaded like any other sample.

3. Select the desired sample and open it.

The sample is loaded and assigned a key zone range that spans the entire keyboard. You can now make settings for it as with any other sample.

Loading REX slices as samples

A slice is a snippet of sound in a REX File. To import a REX slice, click the sample browser button (see above), browse to a REX file and open it as if it was a folder. The browser will then display the slices as files inside that “folder”.

In the rest of this manual, when we refer to importing samples, all that is said applies to REX slices as well.

Creating Key Zones

A “key zone” is a range of keys, that plays a sample. All key zones together make up a “key map”.

To create a new key zone, the following methods can be used:

- **Select “Split Key Zone” from the Edit or context menus.**
This splits the currently selected key zone in the middle. The new zone is the upper half of the split, and is empty. The dividing point has a “handle” above it, see “Setting the Key Zone Range” below for a description.
- **By [Alt]/[Option]-clicking at a point just above the key zone strip, a new empty key zone is created.**
The point where you click becomes the lower limit (or boundary) for the original key zone, and the upper limit for the new key zone.



The new empty key zone gets selected upon creation.

Selecting Key Zones

Only one key zone can be selected at a time. A selected key zone is indicated by a light blue (as opposed to dark blue) strip above the keyboard in the display. There are two ways you can select key zones:

- **By clicking on an unselected key zone in the display.**
- **By activating the “Select Key Zone via MIDI” button.**
Playing a note belonging to an unselected key zone from your MIDI keyboard, will select the key zone it belongs to.

Setting the Key Zone Range

! **Key zones cannot overlap.**

When you adjust the boundaries of a key zone, the surrounding boundaries are automatically adjusted accordingly.

You can change the key zone range in the following ways:

- **By dragging the “handle(s)” which divides the key zones, you can change the range of the selected key zone.**
In the case of having two key zones split in the middle, you could thus change the lower limit for the upper (new) key zone and the upper limit for the original key zone.



- **By using the “Lowkey” and “Highkey” knobs to set a lower and upper range, respectively.**



Deleting a Key Zone

- **To delete a key zone, select it and then select “Delete Key Zone” from the Edit menu.**

About Key zones, Assigned and Unassigned Samples

When you load samples and rearrange your key mapping, you will often end up with samples that are not assigned to any key zone. In the following texts we refer to the samples as follows:

- *Assigned samples* are samples that are currently assigned to one or more key zones.
- *Unassigned samples* are samples that reside in the sample memory, but that are currently not assigned to any key zone.

Adding Sample(s) to a Key Map

If the sample hasn't been loaded yet

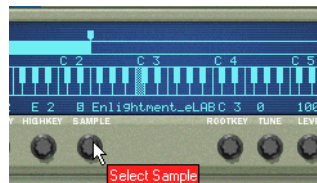
1. **Select a key zone.**
This can be empty, or contain a sample - it doesn't matter for now.
2. **Use the Sample Browser to add one, or several (see below), sample(s).**

The following will happen:

- If the zone contained a sample prior to loading, this will be *replaced*, both in the zone and in the sample memory, unless the sample was also used by *another* key zone, in which case it will be kept.
- If you loaded several samples, one of the samples will be assigned to the key zone, and the other samples will be loaded but remain unassigned.

If the sample is already loaded but unassigned

1. **Select a key zone.**
This can be empty, or contain a sample - it doesn't matter for now.
2. **Use the Sample knob to dial in the sample you want the key zone to play.**



The Sample knob.

Setting the Root Key

Once you have defined a key zone, and added a sample, you should set the root key for the sample.

- **Select the key zone the sample belongs to, and click on the key you wish to set the root key to.**

Which key to select is normally determined by the pitch of the sample. For example if the sample plays a F#2 guitar note, click on F#2.

- ★ **Note that it is possible to select a root key outside the key zone, if required.**

Removing Sample(s) from a Key Map

- **To remove a sample, select the zone it belongs to, and then select "Delete Sample" from the Edit or context menus.**
The sample is removed from the zone and from sample memory.
- **To remove a sample from a key zone/map, *without* removing it from memory, you can either select "No Sample" with the Sample knob for that zone, or simply replace it with another sample in the same way.**

Removing All Unassigned Samples

- **To remove all samples that are not assigned to any key zone, select Delete Unused Samples from the Edit menu.**

Rearranging Samples in a Key Map

There is no specific function for rearranging or trading places between samples and key zones. Simply select a key zone and change the current sample assignment with the Sample knob.

Setting Sample Level

For each key zone you can set a volume level, using the Level button below the display. If the transition between two key zones causes a noticeable level difference, this parameter can be used to balance the levels.

Tuning Samples

Sometimes you might find that the samples you wish to use in a key map are slightly out of tune with *each other*. This parameter allows you to tune each sample in a map by +/- half a semitone.

- **Select the key zone(s) that contains the out of tune sample(s), and use the Tune knob below the keyboard display.**

✦ If all samples originate from different sources, and all or most of them are pitched slightly different (a not uncommon sampling scenario), you could first tune them so that they all match each other, and then, if necessary, use the Sample Pitch controls in the Osc section to tune them globally to the “song” you wish to use the samples in.

→ **Note that if all the samples were slightly out of tune by the same amount in relation to the song you intend to use the samples in, it would be much simpler to use the Sample Pitch controls in the Osc section directly.**

Looping Samples



A sample, unlike the cycles of an oscillator for example, is a finite quantity. There is a sample start and end. To get samples to play for as long as you press down the keys on your keyboard, they need to be *looped*.

For this to work properly, you have to first set up two loop points which determine the part of the sample that will be looped, and make this a part of the audio file. You cannot set loop points in the NN-19, this has to be done in a sample editor.

All included samples already have set loop points (if needed).

For each sample (or key zone), you can select the following Loop modes by using the Loop knob below the keyboard display:

→ **OFF**

No looping is applied to the sample.

→ **FWD**

The part between the loop points plays from start to end, then the cycle is repeated. This is the most common loop mode.

→ **FWD - BW**

The part between the loop points plays from start to end, then from end to start (backwards), and then repeats the cycle.

! **For samples without any loop points, the whole sample will be looped.**

About the Solo Sample Function

The Solo Sample button will allow you to listen to a selected sample over the entire keyboard range.

→ **Select the key zone the sample is assigned to, and then activate Solo Sample.**

This can be useful for checking if the root key is set correctly or if the current range is possible to extend etc.

! **For Solo Sample to work, “Select Key Zone via MIDI” must be disabled!**

Automap Samples

If you have a number of samples that belong together, but haven't mapped them to key zones you can use the "Automap Samples" function on the Edit menu. This is used in the following way:

1. **Select all samples that belong together and load them in one go, using the sample browser.**

One of the samples will be assigned to a key zone spanning the whole range, and the rest will be loaded in to memory but remain unassigned.

2. **Select Automap Samples from the Edit menu.**

Now all samples currently in memory (assigned or unassigned) will be arranged automatically so that:

- **Each sample will be placed correctly according to its root note, and will be tuned according to the information in the sample file.**
Most audio editing programs can save root key information as part of the file.

- **Each sample will occupy half the note range to the next sample's root note.**

The root key will always be in the middle of each zone, with the zone extending both down and up in relation to the root position.

Mapping Samples Without Root Key or Tuning Information

Some samples may not have any information about root key or tuning stored in the file. If the file names indicate the root key you can manually set it for each sample using the method described below. In a worst case scenario, i.e. no tuning or root key information whatsoever, you can still make use of the Automap function:

1. **Select all samples that belong together and load them in one go, using the sample browser.**

One of the samples will be assigned to a key zone spanning the whole range, and the rest will be loaded in to memory but remain unassigned.

2. **Manually set the root key, and adjust the tune knob if the sample needs fine-tuning.**

Without any information stored in the file, or if the file name doesn't indicate the root key, you will have to use your ears for this step. Play the sample and use another instrument or a tuner to determine its pitch.

3. **Select the next sample using the Sample knob, and repeat the previous step.**

Proceed like this until you have set a root key for all the samples in memory.

4. **Select "Automap Samples" from the edit menu.**

The samples will be automatically mapped according to their set root key positions!

How Mapping Information is Saved

All information about key zones, high and low range, root key etc. is stored as part of the Sampler Patch. The original sample files are never altered!

NN-19 Synth Parameters

The NN-19 synth parameters are used to shape and modulate samples. These are mostly similar to the parameters used to shape the oscillators in Subtractor - you have envelope generators, a filter, velocity control etc. Again, it is important to remember that these parameters do not alter the audio files in any way, only the way they will play back.

! **These parameters are global, in the sense that they will affect all samples in a sample patch.**

The Oscillator Section



For a sample patch, the actual samples are what oscillators are for a synthesizer, the main sound *source*. The following settings can be made in the Osc section of the NN-19:

Sample Start

This changes the start position of samples in a sample patch. Turning the knob clockwise gradually offsets the samples' start position, so that they will play back from a position further "into" the samples' waveform. This is useful mainly for two things:

→ **Removing "air" or other unwanted artefacts from the start of less than perfect samples.**

Occasionally (although not in any samples supplied with Reason) you may come across samples where the start point of the sample is slightly ahead of the start of the actual sound. There may be noise or silence in the beginning which was not intended to be part of the sample. By adjusting the sample start position, this can be removed.

→ **Changing the start point as an effect.**

For example, if you had a sample of someone saying "one, two, three", you could change the start position so that when you played the sample it would start on "three".

★ **You can also assign velocity sample start allowing to use your playing to determine the exact sample start. See later in this chapter.**

Setting Sample Pitch - Octave/Semitone/Fine

By adjusting the corresponding knobs you can change the pitch of all samples belonging to a patch, in three ways:

→ **Octave steps**

The range is 0 - 8. The default setting is 4.

→ **Semitone steps**

Allows you to raise the frequency in 12 semitone steps (1 octave).

→ **Fine steps (100th of a semitone)**

The range is -50 to 50 (down or up half a semitone).

! **Note that the controls in this section cannot be used to tune samples against each other, as all samples will be affected equally. To tune individual samples, you use the Tune parameter below the keyboard display (see [page 140](#)).**

Keyboard Tracking

The Osc section has a button named "Kbd. Track". If this is switched off, the sample's pitch will remain constant, regardless of any incoming note pitch messages, although the oscillator still reacts to note on/off messages. This could be useful if you are using non-pitched samples, like drums for example. You could then play a sample in a zone using several keys, allowing for faster note triggering if you wanted to play a drum roll, for example.

Osc Envelope Amount

This parameter determines to what degree the overall pitch of the samples will be affected by the Filter Envelope (see [page 145](#)). You can set negative or positive values here, which determines whether an envelope parameter should raise or lower the pitch.

The Filter Section



Filters are used for shaping the overall timbre of the sound. The filter in NN-19 is a multimode filter with five filter types.

Filter Mode

With this selector you can set the filter to operate as one of five different types of filter. These are as follows:

→ 24 dB Lowpass (LP 24)

Lowpass filters let low frequencies pass and cut out the high frequencies. This filter type has a fairly steep roll-off curve (24dB/Octave). Many classic synthesizers (Minimoog/Prophet 5 etc.) used this filter type.

→ 12 dB Lowpass (LP 12)

This type of lowpass filter is also widely used in classic analog synthesizers (Oberheim, TB-303 etc.). It has a gentler slope (12 dB/Octave), leaving more of the harmonics in the filtered sound compared to the LP 24 filter.

→ Bandpass (BP 12)

A bandpass filter cuts both high and low frequencies, while midrange frequencies are not affected. Each slope in this filter type has a 12 dB/Octave roll-off.

→ High-Pass (HP12)

A highpass filter is the opposite of a lowpass filter, cutting out the lower frequencies and letting the high frequencies pass. The HP filter slope has a 12 dB/Octave roll-off.

→ Notch

A notch filter (or band reject filter) could be described as the opposite of a bandpass filter. It cuts off frequencies in a narrow midrange band, letting the frequencies below and above through.

Filter Frequency

The Filter Frequency parameter (often referred to as “cutoff”) determines which area of the frequency spectrum the filter will operate in. For a lowpass filter, the frequency parameter could be described as governing the “opening” and “closing” of the filter. If the Filter Freq is set to zero, none or only the very lowest frequencies are heard, if set to maximum, all frequencies in the waveform are heard. Gradually changing the Filter Frequency produces the classic synthesizer filter “sweep” sound.

! Note that the Filter Frequency parameter is usually controlled by the Filter Envelope (see “Envelope Section” below) as well. Changing the Filter Frequency with the Freq slider may therefore not produce the expected result.

Resonance

The filter resonance parameter (sometimes called Q) is used to set the Filter characteristic, or quality. For lowpass filters, raising the filter Res value will emphasize the frequencies around the set filter frequency. This produces a generally thinner sound, but with a sharper, more pronounced filter frequency “sweep”. The higher the resonance value, the more resonant the sound becomes until it produces a whistling or ringing sound. If you set a high value for the Res parameter and then vary the filter frequency, this will produce a very distinct sweep, with the ringing sound being very evident at certain frequencies.

- For the highpass filter, the Res parameter operates just like for the lowpass filters.
- When you use the Bandpass or Notch filter, the Resonance setting adjusts the width of the band. When you raise the Resonance, the band where frequencies are let through (Bandpass), or cut (Notch) will become narrower. Generally, the Notch filter produces more musical results using low resonance settings.

Envelope Section

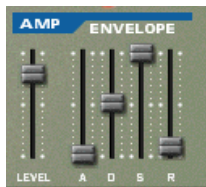
Envelope generators are used to control several important sound parameters in analog synthesizers, such as pitch, volume, filter frequency etc. Envelopes govern how these parameters should respond over time - from the moment a note is struck to the moment it is released.

Standard synthesizer envelope generators have four parameters; Attack, Decay, Sustain and Release (ADSR).

There are two envelope generators in the NN-19, one for volume, and one for the filter frequency.

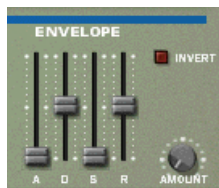
! **Please refer to the Subtractor chapter for a description of the basic envelope parameters.**

Amplitude Envelope



The Amp Envelope is used to adjust how the volume of the sound should change from the time you press a key until the key is released. By setting up a volume envelope you sculpt the sound's basic shape with the four Amplitude Envelope parameters, Attack, Decay, Sustain and Release. This determines the basic character of the sound (soft, long, short etc.). The Level parameter acts as a general volume control for the sample patch.

Filter Envelope



The Filter Envelope can be used to control two parameters; filter frequency and sample pitch. By setting up a filter envelope you control how the filter frequency and/or the sample pitch should change over time with the four Filter Envelope parameters, Attack, Decay, Sustain and Release.

Filter Envelope Amount

This parameter determines to what degree the filter will be affected by the Filter Envelope. Raising this knob's value creates more drastic results. The Envelope Amount parameter and the set filter frequency are related. If the Filter Freq slider is set to around the middle, this means that the moment you press a key the filter is already halfway open. The set Filter Envelope will then open the filter further from this point. The Filter Envelope Amount setting affects *how much* further the filter will open.

Filter Envelope Invert

If this button is activated, the envelope will be inverted. For example, normally the Decay parameter lowers the filter frequency, but after activating Invert it will instead raise it, by the same amount. Note that Invert does not affect the Osc pitch parameter (this can be inverted by setting positive or negative values).

LFO Section



LFO stands for Low Frequency Oscillator. LFOs are oscillators in the sense that they generate a waveform and a frequency. However, there are two significant differences compared to normal sound generating oscillators:

- LFOs only generate waveforms with low frequencies.
- The output of the two LFOs are never actually heard. Instead they are used for modulating various parameters.

The most typical application of an LFO is to modulate the pitch of a (sound generating) oscillator or sample, to produce vibrato.

The LFO section has the following parameters:

Waveform

LFO 1 allows you to select different waveforms for modulating parameters. These are (from the top down):



Waveform	Description
Triangle	This is a smooth waveform, suitable for normal vibrato.
Inverted Sawtooth	This produces a "ramp up" cycle. If applied to an oscillator's frequency, the pitch would sweep up to a set point (governed by the Amount setting), after which the cycle immediately starts over.
Sawtooth	This produces a "ramp down" cycle, the same as above but inverted.
Square	This produces cycles that abruptly changes between two values, usable for trills etc.
Random	Produces random stepped modulation to the destination. Some vintage analog synths called this feature "sample & hold".
Soft Random	The same as above, but with smooth modulation.

Destination

The available LFO Destinations are as follows:

Destination	Description
Osc	Selecting this makes LFO control the pitch (frequency) of the sample patch.
Filter	Selecting this makes the LFO control the filter frequency.
Pan	Selecting this makes the LFO modulate the pan position of samples, i.e. it will move the sound from left to right in the stereo field.

Sync

By clicking this button you activate/deactivate LFO sync. The frequency of the LFO will then be synchronized to the song tempo, in one of 16 possible time-divisions. When sync is activated, the Rate knob (see below) is used for setting the desired timedivision.

Turn the knob and check the tooltip for an indication of the timedivision.



Rate

The Rate knob controls the LFO's frequency. Turn clockwise for a faster modulation rate.

Amount

This parameter determines to what degree the selected parameter destination will be affected by the LFO. Raising this knob's value creates more drastic results.

Play Parameters

This section deals with two things: Parameters that are affected by how you play, and modulation that can be applied manually with standard MIDI keyboard controls.

These are:

- Velocity Control
- Pitch Bend and Modulation Wheel
- Legato
- Portamento
- Polyphony
- Voice Spread
- External Controllers

Velocity Control



Velocity is used to control various parameters according to how hard or soft you play notes on your keyboard. A common application of velocity is to make sounds brighter and louder if you strike the key harder. By using the knobs in this section, you can control how much the various parameters will be affected by velocity. The velocity sensitivity amount can be set to either positive or negative values, with the center position representing no velocity control.

The following parameters can be velocity controlled:

Destination	Description
Amp	This lets you velocity control the overall volume of the sound. If a positive value is set, the volume will increase the harder you strike a key. A negative value inverts this relationship, so that the volume decreases if you play harder, and increases if you play softer. If set to zero, the sound will play at a constant volume, regardless of how hard or soft you play.
F. Env	This sets velocity control for the Filter Envelope Amount parameter. A positive value will increase the envelope amount the harder you play. Negative values invert this relationship.
F. Dec	This sets velocity control for the Filter Envelope Decay parameter. A positive value will increase the Decay time the harder you play. Negative values invert this relationship.
S.Start	This sets velocity control for the Sample Start parameter. A positive value will increase the Start Time amount the harder you play. Negative values invert this relationship.
A. Attack	This sets velocity control for the Amp Envelope Attack parameter. A positive value will increase the Attack time the harder you play. Negative values invert this relationship.

Pitch Bend and Modulation Wheels

The Pitch Bend wheel is used for “bending” notes, like bending the strings on a guitar. The Modulation wheel can be used to apply various modulation while you are playing. Virtually all MIDI keyboards have Pitch Bend and Modulation controls. NN-19 also has two functional wheels that could be used to apply real time modulation and pitch bend should you not have these controllers on your keyboard, or if you aren’t using a keyboard at all. The wheels mirror the movements of the MIDI keyboard controllers.

Pitch Bend Range

The Range parameter sets the amount of pitch bend when the wheel is turned fully up or down. The maximum range is “24” (=up/down 2 Octaves).

Modulation Wheel

The Modulation wheel can be set to simultaneously control a number of parameters. You can set positive or negative values, just like in the Velocity Control section. The following parameters can be affected by the modulation wheel:

Destination	Description
F. Freq	This sets modulation wheel control of the Filter Frequency parameter. A positive value will increase the frequency if the wheel is pushed forward. Negative values invert this relationship.
F. Res	This sets modulation wheel control of the Filter Resonance parameter. A positive value will increase the resonance if the wheel is pushed forward. Negative values invert this relationship.
F. Dec	This sets modulation wheel control for the Filter Envelope Decay parameter. A positive value will increase the decay if the wheel is pushed forward. Negative values invert this relationship.
LFO	This sets modulation wheel control of the LFO Amount parameter. A positive value will increase the Amount if the wheel is pushed forward. Negative values invert this relationship.
Amp	This sets modulation wheel control for the Amp level parameter. A positive value will increase the level if the wheel is pushed forward. Negative values invert this relationship.

Legato

Legato works best with monophonic sounds. Set Polyphony (see below) to 1 and try the following:

- ➔ **Hold down a key and then press another key without releasing the previous.**
Notice that the pitch changes, but the envelopes do not start over. That is, there will be no new “attack”.
- ➔ **If polyphony is set to more voices than 1, Legato will only be applied when all the assigned voices are “used up”.**
For example, if you had a polyphony setting of “4” and you held down a 4 note chord, the next note you played would be Legato. Note, however, that this Legato voice will “steal” one of the voices in the 4 note chord, as all the assigned voices were already used up!

Retrig

This is the “normal” setting for playing polyphonic patches. That is, when you press a key without releasing the previous, the envelopes are retriggered, like when you release all keys and then press a new one. In monophonic mode, Retrig has an additional function; if you press a key, hold it, press a new key and then release that, the first note is also retriggered.

Portamento (Time)

Portamento is when the pitch “glides” between the notes you play, instead of instantly changing the pitch. The Portamento knob is used to set how long it takes for the pitch to glide from one pitch to the next. If you don’t want any Portamento at all, set this knob to zero.

Setting Number of Voices - Polyphony

This determines the polyphony, i.e. the number of voices a patch can play simultaneously. This can be used to make a patch monophonic (=a setting of “1”), or to extend the number of voices available for a patch. The maximum number of voices you can set a patch to use is 99.

! Note that the Polyphony setting does not “hog” voices. For example, if you have a patch that has a polyphony setting of ten voices, but the part the patch plays only uses four voices, this won’t mean that you are “wasting” six voices. In other words, the polyphony setting is not something you need to consider if you want to conserve CPU power - it is only the number of voices *actually used* that counts.

Voice Spread

This parameter can be used to control the stereo (pan) position of voices. The Spread knob determines the intensity of the panning. If this is set to “0”, no panning will take place. The following pan modes can be selected:

Mode	Description
Key	This will shift the pan position gradually from left to right the higher up on the keyboard you play.
Key 2	This will shift the pan position from left to right in 8 steps (1/2 octave) for each consecutive higher note you play, and then repeat the cycle.
Jump	This will alternate the pan position from left to right for each note played.

Low Bandwidth

This will remove some high frequency content from the sound, but often this is not noticeable (this is especially true if you have “filtered down” samples). Activating this mode will save you some extra computer power, if needed.

Controller Section

NN-19 can receive common MIDI controller messages, and route these to various parameters. The following MIDI messages can be received:

- Aftertouch (Channel Pressure)
- Expression Pedal
- Breath Control

If your MIDI keyboard is capable of sending Aftertouch messages, or if you have access to an Expression Pedal or a Breath controller, you can use these to modulate NN-19 parameters. The “Source” selector switch determines which of these message-types should be received.

These messages can then be assigned to control the following parameters:

F. Freq	This sets external modulation control of the filter frequency parameter. A positive value will increase the frequency with higher external modulation values. Negative values invert this relationship.
LFO 1	This sets external modulation control of the LFO Amount parameter. A positive value will increase the LFO amount with higher external modulation values. Negative values invert this relationship.
Amp	This let’s you control the overall volume of the sound with external modulation. If a positive value is set, the volume will increase with higher external modulation values. A negative value inverts this relationship.

Connections

On the back panel of the NN-19 you will find the connectors, which are mostly CV/Gate related.



Audio Outputs

These are the main left and right audio outputs. When you create a new NN-19 device, these are auto-routed to the first available channel on the audio mixer.

Mono Sequencer Control

These are the main CV/Gate inputs. CV controls the note pitch. Gate inputs trigger note on/off values plus a *level*, which can be likened to a velocity value. If you want to control the NN-19 from a Matrix Pattern Sequencer for example, you would normally use these inputs. The inputs are "mono", i.e. they control one voice in the sampler.

Modulation Inputs

! Remember that CV connections will not be stored in the sample patch, even if the connections are to/from the same NN-19 device!

These control voltage (CV) inputs (with associated voltage trim pots), can modulate various NN-19 parameters from other devices, or from the modulation outputs of the same NN-19 device. These inputs can control the following parameters:

- Osc (sample) Pitch
- Filter Cutoff
- Filter Resonance
- Amp Level
- Mod Wheel

Modulation Outputs

The Modulation outputs can be used to voltage control other devices, or other parameters in the same NN-19 device. The Modulation Outputs are:

- Filter Envelope
- LFO

Gate Inputs

These inputs can receive a CV signal to trigger the envelopes. Note that connecting to these inputs will override the "normal" triggering of the envelopes. For example, if you connected a LFO output to the Gate Amp input, you would not trigger the amp envelope by playing notes, as this is now controlled by the LFO. In addition you would only hear the LFO triggering the envelope for the notes that you *hold down*.

- Amp Envelope
- Filter Envelope



REASON

15

→ NN-XT Sampler

Introduction

Features

The basic functions of the NN-Xt are very similar to those of its sampler companion in the Reason rack - the NN-19. Just like the NN-19, NN-Xt lets you load samples and create multi-sample patches by mapping samples across the keyboard. The sound can then be modified by a comprehensive set of synth-type parameters. There are however some major differences between the two. The NN-Xt has:

- **Support for SoundFonts.**

Presets and samples from SoundFont banks can be loaded and used in the NN-Xt (see [page 153](#)).

- **8 stereo output pairs.**

This makes it possible to route different samples to different mixer channels for individual effect processing (see [page 174](#)).

- **The possibility to create layered sounds.**

This is done by mapping several samples across the same keyboard range (see [page 169](#)).

- **The possibility to create sounds that only play over certain velocity ranges, velocity switched key maps and velocity crossfading.**

See [page 170](#).

- **Key maps with individual synth parameter settings for each sample.**

See [page 175](#).

Even though the NN-Xt is a more advanced sample player than NN-19, it should not be considered as a successor to the NN-19, but rather as a complement to it. The NN-19 will for example probably still be the sampler of choice for those of you who want to be able to quickly load a couple of samples and start playing, since that particular aspect takes a little more doing with the NN-Xt.

Panel Overview

The Main Panel

When the NN-Xt is added to the rack, you will initially only see the main panel.

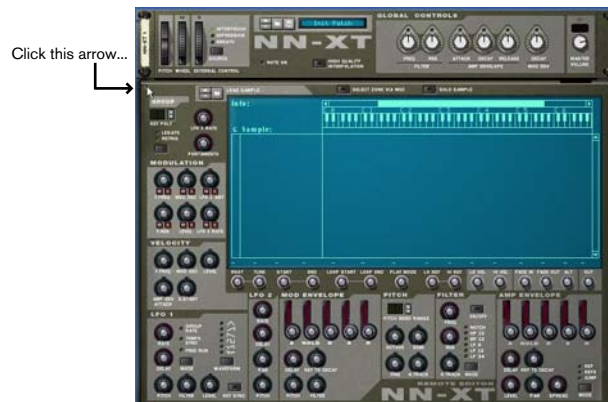


The NN-Xt main panel.

The main panel is where you load complete sample patches. It also contains the "global controls". These are controls that affect and modify the sound of entire patches rather than the individual key zones.

The Remote Editor panel

To show/hide the remote editor panel, use the fold/unfold arrow at the bottom left.



...to make the remote editor panel visible.

The remote editor panel is where you load individual samples, create key maps, modify the sound of the samples with synth parameters etc.

! The main panel of the NN-Xt can be folded like any other Reason device. Note that folding the main panel will also fold the remote editor regardless of its current state.

Loading Complete Patches and REX Files

As previously alluded, you can load complete sample patches as well as individual samples into the NN-XT.

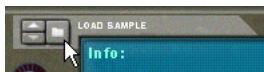
- A patch is a complete “sound package”. It contains information about all the samples used, assigned key zones, associated panel settings etc. Loading a sample patch is done by using the patch browser on the main panel, and works in the same way as with any other Reason device.



The patch browser button on the main panel.

For general instructions on how to load and save patches, please consult the chapter “Working with Patches” in the Getting Started book.

- Loading separate samples is done in a similar way, but via the sample browser on the remote editor panel. If you load samples, map them across keyboard ranges and set up the sound the way you want it, you can save your settings as a Patch for easy access later.



The sample browser button on the remote editor.

More about loading samples later in this chapter.

Loading NN-XT Patches

NN-XT Patches are patches made specifically for the NN-XT. Reason ships with a large number of NN-XT Patches, some in the Factory Sound Bank but most in the Orkester Sound Bank. NN-XT Patches have the extension “.sxt”.

Loading NN-19 Patches

NN-19 Patches have the extension “.smp”. Note that when loading NN-19 patches into the NN-XT, some parameters will not be applicable since the NN-19 and the NN-XT to some extent differ from each other in terms of controls. In these cases, the concerned parameters will either be ignored by the NN-XT or mapped to the most equivalent control.

Loading SoundFonts

The SoundFont format was developed by E-mu systems in collaboration with Creative Technologies. It is a standardized data format containing wavetable synthesized audio and information on how it should be played back in wavetable synthesizers - typically on audio cards. The SoundFont format is an open standard so there is a vast amount of SoundFont banks and SoundFont compatible banks developed by third parties.

Loading SoundFonts is no different from loading NN-XT Patches. As with NN-19 Patches, the NN-XT does its best to map all the SoundFont settings to NN-XT parameters.

You can load SoundFont *presets* by using the patch browser, and single SoundFont *samples* by using the sample browser.

Loading Complete REX Files as Patches

REX Files are files created in the ReCycle program. This is an application created by Propellerhead Software, used for slicing up music loops and enabling them to be played back in any tempo. In Reason, REX files are primarily used in the Dr. Rex loop player, but they can be used in the NN-XT as well. Possible extensions are “.rx2”, “.rcy” and “.rex”.

When loading a REX file, each slice in the file is assigned to one key, chromatically. All parameters are set to default settings.

When using REX files in the DR. Rex loop player, it is possible to make a track play the slices in order to recreate the original loop. To do the same in the NN-XT requires a few extra steps.

1. **Use the patch browser to load the REX file into an NN-XT sampler.**
2. **Create a Dr. Rex loop player and load the same REX file into this device.**
3. **Use the To Track feature on the Dr. Rex to create playback data (a group) on the track assigned to the Dr. Rex.**
4. **Move that group to the track that plays the NN-XT and play it back from there.**
5. **Delete the Dr. Rex loop player.**

Using the Main Panel

All of the controls on the main panel are used for globally modifying certain parameters for *all* of the samples in a patch, by the same amount.

! Movements of the parameters on the main panel can be recorded as automation. However, controls on the remote editor panel (described later) can not!

The following is a description of the controls and parameters on the main panel.

The Pitch Bend and Modulation wheels



Most MIDI keyboards come equipped with Pitch Bend and Modulation wheels. The NN-XT features settings for how incoming MIDI Pitch Bend and Modulation wheel messages should affect the sound. The wheels on the NN-XT will also mirror the movements of the wheels on your MIDI keyboard.

If you don't have Pitch Bend or Modulation controls on your keyboard, or if you aren't using a keyboard at all, you can use the two fully functional wheels on the NN-XT to apply real time modulation and pitch bend.

- The Pitch Bend wheel is used for “bending” the played notes up and down to change their pitch - much like bending the strings on a guitar or other string instrument. The Pitch Bend Range is set on the remote editor panel (see [page 178](#)).
- The Modulation wheel can be used for applying modulation to the sound while you're playing. It can also be used for controlling a number of other parameters, as described on [page 175](#).

External control



This section can be used in three ways:

Receiving MIDI controller messages from external sources

NN-XT can receive common MIDI controller messages, and route these to various parameters. You use the “Source” selector switch to determine which type of message should be received:

- Aftertouch (Channel Pressure)
- Expression Pedal
- Breath Control

If your MIDI keyboard is capable of sending aftertouch messages, and/or if you have connected an expression pedal or a breath controller to it, you can use these to modulate NN-XT parameters. Which parameters should be modulated is set in the remote editor panel (see [page 175](#)).

Recording MIDI controller messages with the wheel

The wheel in the external control section can be used for recording any or all of the three MIDI controller message types into the Reason sequencer. If your MIDI keyboard isn't capable of sending aftertouch messages or you don't have access to an expression pedal or a breath controller, you can use the wheel instead.

This is done just as with any other automation recording, see [page 8](#).

High Quality Interpolation



This switch turns High Quality Interpolation on and off. When it is activated, the sample pitch is calculated using a more advanced interpolation algorithm. This results in better audio quality, especially for samples with a lot of high frequency content.

- High Quality Interpolation uses more computer power however - so if you don't need it, it's a good idea to turn it off! Listen to the sounds in a context and determine whether you think this setting makes any difference.
- If you are using a Macintosh with a G4 (Altivec) processor, High Quality Interpolation is always activated, regardless of the state of this button.

Global Controls



All of these knobs change the values of various parameters in the remote editor panel and affect *all* loaded samples. Thus they can be used for quickly adjusting the overall sound.

The knobs are bi-polar, which means that when they are centered, no parameter change is applied. By turning them to the right you increase the corresponding value, and by turning them to the left, you decrease the value.

Again, the movements of these parameters can be recorded as automation. This is done just as with any other automation recording, see [page 8](#).

The controls are, from left to right:

Filter

These two knobs each control a parameter of the filter (see [page 179](#)). Note that the filter must be on for these to have any effect.

→ Frequency

This changes the cutoff frequency of the filter.

→ Resonance

This changes the resonance parameter of the filter, meaning - the filter characteristic, or quality.

Amp Envelope

These three knobs control the Amplitude Envelope (see [page 181](#)) in the following way:

→ Attack

This changes the Attack value of the Amplitude Envelope. That is, how long it should take for the sound to reach full level after you press a key on your keyboard.

→ Decay

This changes the Decay value of the Amplitude Envelope. Decay determines how long it should take for the sound to go back to the sustain level after it has reached full value (see [page 181](#)) and the key that triggered the sound is still being pressed.

→ Release

This changes the Release value of the Amplitude Envelope. Release works just like Decay with the exception that it determines how long it should take for the sound to become silent after the key has been *released*.

Mod Envelope

This knob controls the Decay value of the Modulation Envelope (see [page 180](#)). Also see above for a brief description of Decay.

Master Volume

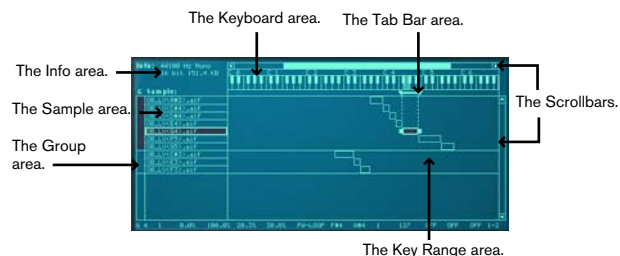
This controls the main volume out from the NN-Xt. Turn the knob to the right to increase the volume.

Overview of the Remote Editor panel

It is in the Remote Editor Panel that the main NN-XT action is going on, especially if you're creating your own patches. The remote editor is dominated by the key map display, and this is also the part on which we will concentrate to begin with.

The Key Map display

The key map display consists of a number of separate areas that let you do different things. To help you navigate the key map display, these areas are described below.



The Info area

This displays the following information about the currently selected sample: Sample rate, mono/stereo information, bit resolution and file size.

The Sample area

This area displays the names of the samples in each zone. It also allows you to change the order of the zones by clicking and dragging them up and down.

The Group area

This area does not show any information. However, by clicking in it, you can instantly select all the zones that belong to a certain group. See [page 163](#) for information on how to create groups.

The Keyboard area

Aside from the fact that it is a guideline for setting up key ranges, it is also used for setting the root keys of, and auditioning loaded samples. See [page 167](#) and [page 162](#) respectively for more information.

The Tab Bar area

This area gives you a visual indication of the key range of a selected zone. By clicking and dragging the "handles" at the key range boundaries, you can resize the key ranges, and by clicking in between the handles, you can move the key ranges without changing their length.

The Key Range area

This area in the middle of the key map display is where you keep track of all the zones and the relationship between them. You can also move and resize the zones just like in the Tab Bar area, as described above.

The Scrollbars

There are both horizontal and vertical scrollbars that work just like regular scrollbars. Whenever there is more information in the key map display than what fits on a "single screen", you can use the scrollbars to reveal it. Either click on the arrows or click and drag the scrollbar handles.

Sample Parameters



This area shows the current values of basic parameters you can set for each separate zone, such as root key, play mode, output etc. The parameters are changed by using the knobs directly below the key map display.

Group Parameters



These parameters are adjusted on a per group basis (see [page 174](#) for more information on groups). Most of them relate to performance or playing style.

Synth Parameters



The bulk of the parameters on the remote editor are used for adjusting the sound of the samples by applying filtering, envelope shaping, modulation (like vibrato and tremolo) and so on. We call these the synth parameters, since they are to a large extent identical to those on a regular synthesizer.

About Samples and Zones

For a clear understanding of the terminology used when describing the various operations that can be performed in the key map display, it is important to clarify the distinction between a *sample* and a *zone*:

- A Sample is a piece of audio that can be loaded into the NN-XT and played back.
- A Zone could be viewed as a “container” for a loaded sample.

All loaded samples are placed in “Zones” in the key map display. You can then organize the zones as you please, and make various settings such as key- and velocity ranges separately for each zone.

In other words, the settings you make are actually performed on the zones, but affect the samples in them. Hence, when we talk about making settings for a zone, it is synonymous with making settings for a sample - the sample that the zone contains.

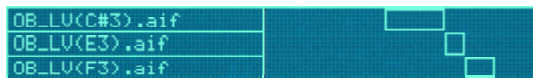
- Two or more zones can play the same sample, but with different parameter settings, making them sound completely different.
- A zone can be empty, playing no sample at all.

Selections and Edit Focus

Almost all operations in the remote editor are performed on one or more *selected zones* or on the zone with *edit focus*. Several zones can be selected at once, but only *one* zone at a time can have edit focus.

This is important since:

- **Editing operations that can be performed on several zones (like deleting), always apply to the selected zones.**
- **Editing operations that can be performed on one zone only (like adjusting the “Lo key”), always apply to the zone with edit focus.**
- **The front panel always shows the settings for the zone with edit focus.**



Here no zone is selected.



Here the middle zone is selected but does not have edit focus.



Here the middle zone has edit focus but is not selected. Notice the thicker border and the additional handles in the key range area.



Here the middle zone is selected and has edit focus.



Here, all three zones are selected, but the middle one has edit focus.

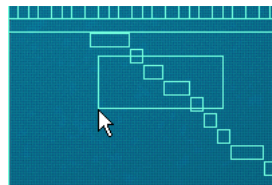
Selecting Zones

- **To select a zone, click on it.**
Clicking on a zone will also automatically give it edit focus.

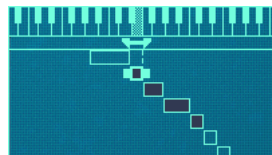
You can also select multiple zones in several ways:

- **By holding down [Shift] or [Command] (Mac)/[Ctrl] (Windows) and clicking on the zones you wish to select.**
This way you can select several non-contiguous zones. You can also deselect a selected zone by clicking on it again.
- **By using the keyboard command [Command]-[A] (Mac)/[Ctrl]-[A] (Windows).**
This will select *all* of the zones in the key map display. To deselect all zones, click in an unoccupied area in the Sample column or the key map area.
- **By clicking and dragging a selection box in the key map area.**

Making a selection box like this....



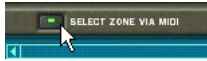
...will select these zones:



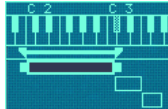
Note that the zones don't have to be completely encompassed by the selection box. The selection box only have to intersect parts of the zones to include them in the selection.

Selecting zones via MIDI

You can also select zones via your MIDI keyboard. By clicking the button marked "Select zones via MIDI" above the key map display so that it lights up, you enable selection via MIDI.



This way, you can select a zone and give it edit focus by pressing a key that lies within the zone's key range (see later in this chapter for information about setting up key ranges).



In this case, this zone can be selected by pressing any key between C2 - C3 on your MIDI keyboard.

Note also, that selection via MIDI is velocity sensitive. Zones may have specific velocity ranges. This means that they won't be played unless the key that triggers the zone is played with a certain velocity. The same rules apply when selecting via MIDI, only zones that meet the velocity criteria will be selected. Read more about setting up velocity ranges on [page 170](#).

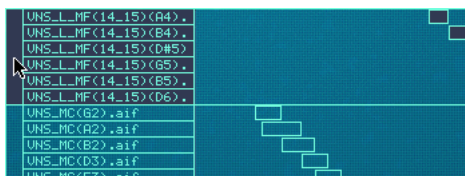
Selecting All Zones in a Group

The concept of zone groups is fully introduced on [page 163](#). For now we will only describe how to select all samples that belong to the same group:

Clicking in the Group column...



...selects all zones in the group



Moving Edit Focus

Moving Edit Focus

A zone can be given edit focus independently of selection:

- When you click on an unselected zone, it both gets selected and gets edit focus.
- When you select several zones using [Shift] or [Command]/[Ctrl], the one you select last always gets edit focus.
- To set edit focus to a zone when several zones are already selected, click on it without holding down any modifier keys.

This way, you can move edit focus between the selected zones freely without deselecting any of them.

Adjusting Parameters

Adjusting Synth Parameters

The synth parameters are the ones that occupy the bulk of the remote editor panel (see [page 157](#)). Changes you make to synth parameters always apply to all selected zones.

- **The panel “only” shows the settings for the zone with edit focus.**
More about this below.
- **To make adjustments to one zone, select it (which also gives it edit focus) and adjust the parameter on the front panel.**
- **To set several zones to the same value, select them and adjust the parameter.**
All zones will be set to the same value for the parameter you adjusted.

Adjusting Group Parameters

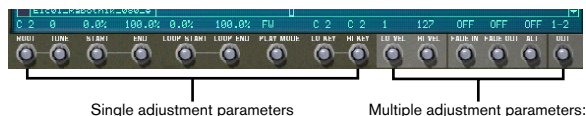
Group parameters apply to a group. That is, they are settings that are shared by all zones in a group.

- **To make adjustments to one group, select one or more zones that belong to the group, and adjust the parameter on the front panel.**
- **To set several groups to the same value, select at least one zone in each group you want to adjust, and adjust the parameter.**
All groups will be set to the same value. More about this below.

Sample Parameters

About Single and Multiple Parameters

The Sample parameters are divided into two groups: single and multiple, differentiated by color on the front panel:



- **Single adjustment parameters can only be applied to one zone at a time. Adjustments made to these parameters always only apply to the zone with *edit focus*.**
- **Multiple adjustment parameters apply to all selected zones, just like the synth parameters, as described above.**

About “Conflicting” Parameters

Often you will find yourself in a situation where you select multiple zones and parameter settings differ between them. This is quite normal. For example, you will often find yourself making adjustments to for example level and filtering to balance the sound between several samples across the keyboard. However, if you have multiple selections this can sometimes lead to confusion: Enter the NN-XT’s “conflicting parameters” indication:

Whenever two or more *selected* zones have conflicting parameter settings, NN-XT will notify you about this by showing a small “M” (for multiple) symbol, next to the parameter.



In this example, Level and Spread have conflicting settings.

- **The controls on the panel always show the setting for the zone with edit focus.**
- **By clicking your way through the zones within the selection, you can see the settings for each zone.**
- **If you adjust a parameter, all selected zones will be set to the same value for this parameter.**

You can put this functionality to good use when checking how a patch has been created and when checking that your own settings are consistent through the various zones.

Copying parameters between zones

You can easily copy parameter settings from one zone to any number of other zones. Proceed as follows:

- Select all the zones you want to involve in the operation.**
By this we mean the zone with the settings you wish to copy, and the zone(s) to which you want to copy the settings.
 - Make sure the zone that contains the settings you want to copy has edit focus.**
 - Pull down the Edit menu or the NN-XT context menu and select “Copy Parameters to Selected Zones”.**
All the selected zones will now get the exact same parameter settings.
- ! **Observe that this only applies to the synth parameters (see [page 175](#)). Sample parameters (root key, velocity range etc.) can not be copied.**

Managing Zones and Samples

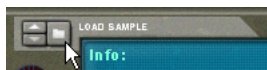
Creating a Key Map

When you add an NN-Xt sampler to the rack, its key map display is always empty. That is, it contains no samples.

To create a new key map, proceed as follows:

1. **Either click the Browse Samples button, select Browse Samples from the Edit menu or select Browse Samples from the NN-Xt's context menu.**

This will bring up the regular Reason file browser.



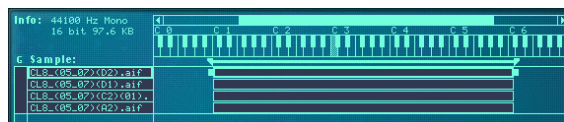
The Browse Samples button.

2. **Select the sample or samples that you want to load in the browser and click "OK".**

The selected sample(s) are loaded into the NN-Xt.

When new samples are loaded into the NN-Xt they have the following properties:

- Each sample is placed in its own zone.
- Each zone spans a key range of five octaves on the keyboard - C1 to C6.
- All the newly added sample(s)/zones are automatically selected.
- The first added zone gets edit focus.



A key map with four newly added samples.

Setting Root Notes and Key Ranges

The next step after loading the samples is most likely to adjust the key range, root note and tuning of the samples, so that they play sensibly across the key range. There are many ways of doing this, described on [page 164](#) and onwards. However, we will here briefly describe a procedure for quickly creating a complete key map out of a set of loaded samples.

This example assumes that the samples you load is a set of multisamples for a pitched instrument (like guitar, piano, flute etc.).

1. **Load the samples.**
2. **Use "Select All" on the Edit menu to select all the loaded samples.**
3. **Use "Set Root Notes from Pitch Detection" to automatically set up the root notes (pitches) for the samples.**
4. **Select "Automap Zones" from the Edit menu.**

All the selected zones will now be arranged automatically into a basic key map. You can now proceed with adjusting the synth parameters on the front panel to shape the sound!

About File Formats and REX Slices

The NN-Xt can import various types of samples:

→ Standard Wave files

These have the extension ".wav". This is the standard audio file format for the PC platform. Any audio or sample editor, regardless of platform, can read and create audio files in Wave format. Any sample rate and practically any bit resolution is supported.

→ Standard AIFF files

These have the extension ".aif" and this is the standard audio file format for the Mac platform. Again, any audio editor can read and create audio files in this format. Any sample rate and practically any bit resolution is supported.

→ SoundFont samples

This is a standardized data format containing wavetable synthesized audio and information on how it should be played back in wavetable synthesizers - typically on audio cards. SoundFont banks are hierarchically organized into different categories: User Samples, Instruments, Presets etc. The NN-Xt lets you load single samples from within a Soundfont bank.

→ REX file slices

A slice is a snippet of sound in a REX File (see [page 153](#)). To import a REX slice, browse to a REX file and open it as if it was a folder. The browser will then display the slices as files inside that "folder". In the rest of this manual, when we refer to importing samples, all that is said applies to REX slices as well.

- ! **If you want files to play back with their original bit depth - if higher than 16-bits - make sure to activate "Use High Resolution Samples" on the General page in the Preferences dialog. Otherwise, samples will be played back as 16-bit files in NN-Xt regardless of their original bit depth. See the Getting Started book for more details.**

Adding More Samples to the Key Map

You can add additional samples to an existing key map in the same way as described above.

1. **Make sure that no already loaded sample has edit focus.**
If you don't, there's a risk that the selected sample will be replaced, see below. To remove the edit focus, click in an unoccupied area in the Sample column or the key map area.
2. **Open the Sample Browser.**
3. **Select the sample(s) you want to load in the browser and click "OK".**
The new sample(s) are added to the key map.

Replacing a Sample

To replace the sample in a zone, proceed as follows:

1. **Make sure the zone has edit focus and do one of the following:**
 - Click the **Browse Samples** button.
 - Select **Browse Samples from the Edit menu or the NN-XT context menu.**
 - **Double click in the zone.**
Any of these methods will open the standard file browser in which you can select a new sample for the zone.
2. **Select one and only one sample in the Sample Browser.**
If you select more than one sample in the browser the samples you load will not replace the one with edit focus. They will instead be added below it.

Quick Browsing through Samples

If you want to quickly browse through a number of samples, for example to see which one of them would fit best in a certain context, proceed as follows:

1. **Set up the zone as desired and make sure it has edit focus:**
2. **Use the arrow buttons in the Browse Samples section to select the next/previous sample in the same directory.**

Removing Samples

- **To remove a sample from a zone, select it by clicking on it and then select "Remove Samples" from the Edit menu or the NN-XT context menu.**
This will remove the sample from the zone, leaving it empty. Note that you can remove the samples from several selected zones at the same time.

Auditioning samples

You can audition the loaded samples in two ways:

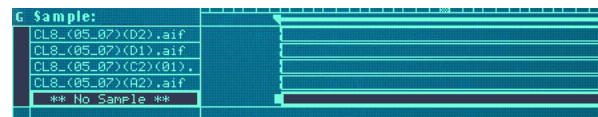
- **By pressing [Option] (Mac)/[Alt] (Windows) and clicking a sample in the sample column.**
The mouse pointer will take on the shape of a speaker symbol when you move it over the sample column.
Clicking a sample will play it back at its root pitch (see [page 167](#)). Furthermore, the sample will play back in its *unprocessed* state. That is, without any synth-parameters applied (see [page 175](#)).
- **By pressing [Option] (Mac)/[Alt] (Windows) and clicking a sample in the keyboard column.**
The difference here is that you will hear the sample at the pitch corresponding to the key you clicked and with any and all processing applied. The click mimics a key played with velocity 100. Also note that this may trigger several samples, depending on whether they are mapped across the same or overlapping key ranges, and the velocity range settings (see [page 164](#) and [page 170](#) respectively).

Adding Empty Zones

You can add empty zones to a key map. Empty zones are treated just like zones containing samples, in that they are automatically selected, gets edit focus and are assigned a five octave key range when they are first created. However, you can only add one zone at a time. It is also possible to resize, move and edit empty zones in the same way as zones containing samples.

- **To add an empty zone, pull down the Edit menu or the NN-XT context menu and select "Add Zone".**

An empty zone is added below any existing zones in the key map. An empty zone is indicated with the text *****No Sample*****.



A newly added empty zone.

After you have added an empty zone, you can assign a sample to it, just as when Replacing a Sample, or when Quick Browsing, as described above.

Duplicating Zones

You can duplicate any number of already existing zones (containing samples or empty).

1. **Select the zone(s) you want to copy.**
2. **Pull down the edit menu or the NN-XT context menu and select “Duplicate Zones”.**

The selected zones will now be copied and automatically inserted below the last one in the key map display.

The duplicated zones will contain references to the same samples as the original zones. They will also have the exact same key ranges and parameter settings.

Using Copy and Paste

The Copy Zones function on the Edit menu allows you to copy all selected zones to the clipboard. Selecting Paste Zones from the Edit menu will paste the zones into the selected NN-XT device, below the existing zones.

This is a handy way to transfer zones (complete with all settings) from one NN-XT device to another.

Removing Zones

To remove one or several zones, select them and do one of the following:

- **Press [Delete] or [Backspace] on the computer keyboard.**
- **Select “Delete Zones” from the Edit menu or the NN-XT context menu.**

When removing zones, you will remove any samples in them as well.

Rearranging Zones in the List

- **To move a zone to another position in the list, click on it in the samples column and drag up or down.**

An outline shows you where the zone will appear when you release the mouse button.

Working with Grouping

About Groups

Grouping has two purposes:

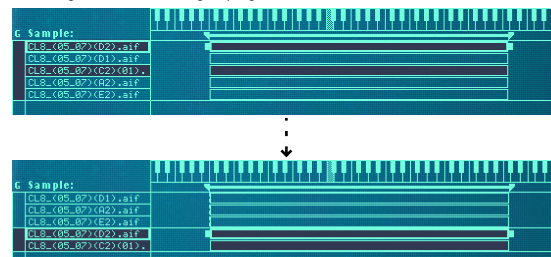
- **To allow you to quickly select a number of zones that “belong together.”**
For example if you have created a layered sound consisting of piano and strings, you could put all string samples in one group and all piano samples in one group. Then you can quickly select all piano samples and make an adjustment to them by trimming a parameter.
- **To group zones that need to share group settings together.**
For example, you may want to set a group to legato and monophonic mode and add some portamento so that you can play a part where you slide between notes.

Note that there is always at least one group, since the zones you create are always grouped together by default.

Creating a Group

1. **Select the zones you want to group together.**
The zones don't have to be contiguous in order to be grouped. Regardless of their original positions in the samples column, they will all be put together in succession.
2. **Select “Group Selected Zones” from the Edit menu or the NN-XT context menu.**
The zones are grouped.

Selecting these zones and grouping them...

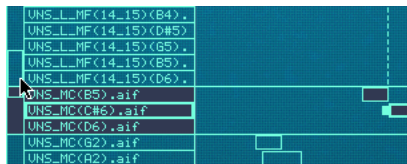


...will create these two groups instead of the original one large group.

Moving a Group to another Position in the List

- Click on the group in the Groups column and drag up or down with the mouse button pressed.

An outline of the group you move is superimposed upon the display to help you navigate to the desired position.



Dragging a group to a new position.

3. Release the mouse button at the desired position.

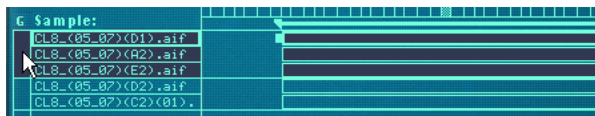
The group and all its zones appear at the new position.

Moving a Zone from one Group to another

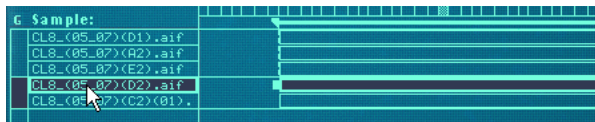
This is done just as when rearranging samples in the list, as described on the previous page. The only difference is that you drag the zone from one group to another.

Selecting a Group and/or Zones in a Group

- Clicking on a group in the groups column selects the group and all the zones in the group.



- Clicking on a zone in the samples column selects the group (and that zone).



The Group Parameters

There are a few parameters on the front panel that apply specifically to groups. see [page 174](#) for details.

Working with Key Ranges

About Key Ranges

Each zone can have its own separate key range, the lowest and the highest key that will trigger the sample.

A good example of use for this is when sampling a certain instrument. Sampling of a piano for example is usually performed by making several recordings of different notes at close intervals, and then mapping these samples to separate, contiguous, fairly narrow key ranges. This concept is called multi-sampling.

The reason for this is that if one single sample is played across the entire keyboard, it will most likely sound very unnatural when played too far from its original pitch, since the amount you can transpose a sound without negatively affecting its timbre is very limited.

Setting up Key Ranges

You can adjust the key range of zones in a number of ways:

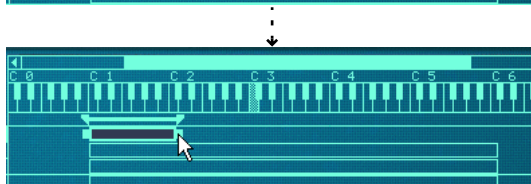
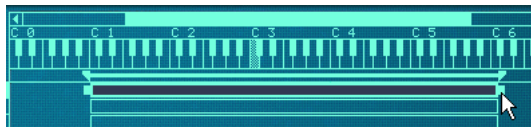
By Dragging the Zone Boundary Handles

1. Select the zone in the Key Range area.
2. Point and click on one of the handles that appear at each end.
3. Drag the handle left/right.

Dotted lines extend from the edges of the zones up to the keyboard area.

These lines give you a visual indication of which keys the key range will encompass. There is also an alphanumeric indication at the bottom left of the display.

Clicking and dragging the high key boundary handle of a zone with the default key range of C1 - C6...



...to change the key range to C1 - C2.

4. Repeat the procedure with as many zones as you wish, to create a complete key map.

By using the Lo Key and Hi Key controls

Directly below the key map area is a row of knobs. These are the sample parameters. As the name implies, they are used for changing various parameters that affect how the zones are played back. In the middle of the sample parameters area are two knobs called “Lo Key” and “Hi Key”.

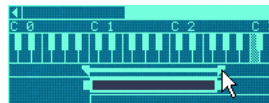


These can be used for setting the low key and the high key of a zone's key range, just like dragging the boundary handles as described above:

1. **Make sure the zone for which you want to set the key range has edit focus.**
2. **Use the knobs to change the corresponding key - low or high.**
Check the display right above the knobs for an indication of the key. You can also keep an eye on the lines extending from the zone edges to the keyboard area.

By Dragging the Zone Boundary Handles on the Tab Bar

As previously described, the area directly below the keyboard area is called the tab bar. This shows the key range for the currently selected zone, and also contains boundary handles.



Dragging a boundary handle on the tab bar.

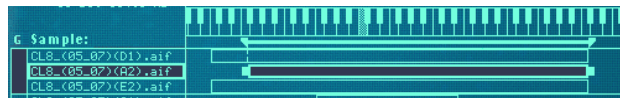
These handles can be used much to the same effect as when dragging the boundary handles in the key map display. However, the handles on the tab bar can change the key range of multiple zones at the same time.

The following applies:

- The tab bar shows the key range for the zone with edit focus.
 - Dragging the boundary handles for that zone will also simultaneously change the key range for a number of surrounding zones *if*:
- **The high key or low key (depending on which handle you drag) of the other zones are the same as the zone with edit focus.**

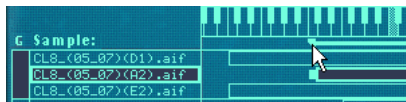
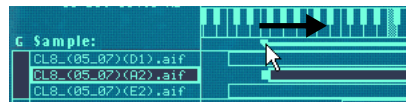
→ The other zones are adjacent to the zone with edit focus.

! **Note that it doesn't matter whether the other zones are selected or not. They will be affected anyway.**

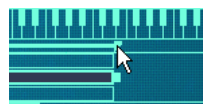
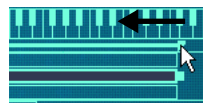


In the example in the picture above, the zone within the middle has edit focus. Its left handle (the low key) is placed differently from any of the other zones, but *all* of the zones have the same high key setting. This means that...

- Dragging the left handle will *only* move the low key position of the zone with edit focus (the pictures show before and after dragging).



- Dragging the *right* handle will move the high key position for *all* of the zones at the same time, since they all have the same high key position (again, the picture shows before and after dragging).



Moving Zones by Dragging the Zone Boxes

You can also move entire zones horizontally, thereby changing their key ranges.

- 1. Select all the zones you want to move.**

You can move several zones simultaneously.

- 2. Point on any of the selected zones, and press the mouse button.**

- 3. Drag left/right and release the mouse button.**

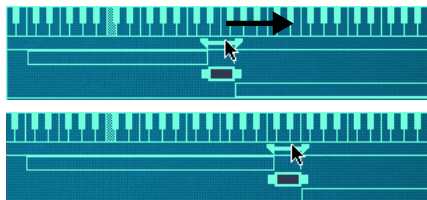


Dragging multiple zones.

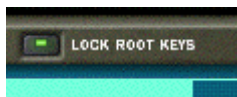
Moving Zones by Dragging in the Tab Bar

You can also move a zone by dragging anywhere between the zone boundary handles on the tab bar. When you do, the surrounding zones will be affected just as when dragging the boundary handles in the tab bar (see above).

This can be used to "slide" a zone in relation to its surrounding zones, as the picture example below shows (before and after dragging).



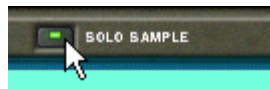
About the Lock Root Keys function



Normally, when you move zones (as described above), the root note of the zone(s) you move will change accordingly. In other words, the zone(s) will be transposed. If this is not desired, you can activate the Lock Root Keys function prior to moving the zone(s) by clicking on the button above the key map display.

Moving zones without changing their root notes can be used for some interesting effects, since it will completely change the timbre of the sample(s) as they are played back.

About the Solo Sample function



The Solo Sample function lets you play a selected sample over the entire keyboard and disregarding any velocity range assigned to the sample. All other loaded samples are temporarily muted.

This is useful if you for example want to check how far up and down from its root key a sample can be played on the keyboard before starting to sound "unnatural". The solo sample function can therefore be useful as a guide for setting up key ranges, as described on [page 164](#).

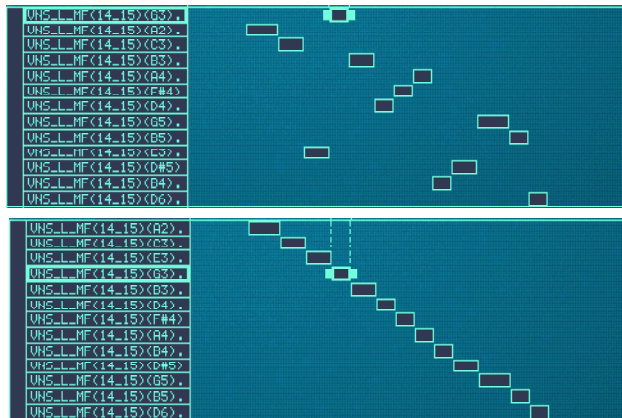
- 1. Select one and only one zone, or - if you have a selection of multiple zones - make sure the one you want to hear has edit focus.**
- 2. Activate Solo Sample by clicking on the button so that it lights up.**
- 3. Play the MIDI keyboard**

Sorting Zones by Note

The Edit menu and the NN-XT context menu contains an item called “Sort Zones by Note”. This option lets you automatically sort the selected zones in descending order according to their key ranges.

When you invoke this option, the selected zones will be sorted from top to bottom in the display starting with the one with the lowest range.

Note however, that the sorting is done strictly on a *group* basis. That is, only zones that belong to the same group can be sorted in relation to each other.



Before sorting and after.

If two zones have the same key range, they are sorted by velocity range.

Setting Root Notes and Tuning

About the Root Key

All instrument sounds have an inherent pitch. When playing a sample of such a sound on the keyboard, the keys you play must correspond to that pitch. For example, you may have recorded a piano playing the key “C3”. When you map this onto the NN-XT key map, you must set things up so that the sampler plays back the sample at original pitch when you press the key C3.

This is done by adjusting the root note.

- **Many samples files from different sources already have a set root key in the file. If they do, the root key will be correctly set automatically when you load the sample into a zone.**
- **However if the sample doesn't have a root note stored in the file, (if you for example have recorded it yourself) you will need to adjust it**

Setting the Root Note Manually

To set the root key for a zone, proceed as follows:

- **Make sure the zone has edit focus (for example by clicking on it), and do one of the following:**
- **Use the knob marked “Root” in the sample parameter area below the display.**

Turning it to the right will raise the pitch of the root key. The selected key is displayed alphanumerically directly above the knob, and you can also look at the keyboard area for a visual indication (see below).



- **Press [Ctrl] (Windows)/[Command] (Mac) and click on the desired root key in the keyboard area.**

The set root key is shaded so you can easily distinguish it.



Tuning Samples Manually

In addition to setting the root note, you may need to fine tune your samples, in order for them to match other instruments and/or each other:

- **Make sure the zone has edit focus (for example by clicking on it).**
- **Use the knob marked “Tune” in the sample parameter area.**
This allows you to tune each sample in a key map by +/- half a semitone (-50 – 0 – 50).



Setting the Root Note and Tuning Using Pitch Detection

The NN-XT features a pitch detection function to help you set the root keys. This is useful if you for example load a sample that you haven't recorded yourself, and you don't have any information about its original pitch.

Proceed as follows:

1. **Select all the zones you want to be subject to pitch detection.**
 2. **Pull down the Edit menu or the NN-XT context menu and select “Set Root Notes from Pitch Detection”.**
The samples in all the selected zones will now be analyzed, and the detected root keys will automatically be set for you.
- ! **Note that for this to work properly, the samples must have some form of perceivable pitch. If it is sampled speech, or a snare drum for example, it probably doesn't have any discernible pitch.**

About Changing the Pitch of Samples

The procedures above should be used to make sure the samples are consistently pitched across the keyboard, and that they all match an absolute reference (for example A 440 tuning).

If you need to tune the samples to match other material, or to get a certain effect (for example detuning two sounds against each other for a chorus effect) you should use the Pitch section among the synth parameters, not the sample tuning parameters.

Using Automap

The automap function can be used as a quick way of creating a key map, or as a good starting point for further adjustments of a key map.

Automap works under the assumption that you intend to create a key map for a complete instrument, for example a number of samples of a piano, all at different pitches.

1. Load the samples you want to Automap.

Now you have three options:

- **Trust that the root note information in the files is already correct.**
- **Manually adjust the root notes (and tuning) for all the samples.**
- **Use “Set Root Notes from Pitch Detection” to automatically set up the root notes.**

2. Select all zones you want to automap.

3. Select Automap Zones from the Edit menu or the NN-XT context menu.

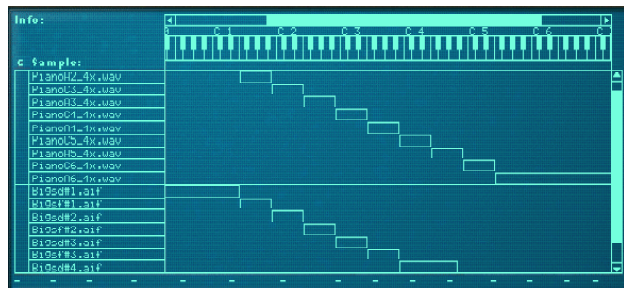
All the selected zones will now be arranged automatically in the following way:

- **The zones will be sorted in the display (from top to bottom - lowest key first) according to the root keys.**
- **The zones will be assigned key ranges according to the root keys.**
The key ranges are set up so that the split between two zones is exactly in the middle between the zones' root notes. If two zones have the same root key they will be assigned the same key range.

Layered, Crossfaded and Velocity Switched Sounds

Creating Layered Sounds

You can set things up so that two or more zones have overlapping key ranges - either completely or partially. This way you can create layered sounds, i.e. different samples that are played simultaneously when you press a key on your keyboard.



In the picture above, you can see a set of piano samples at the top, mapped across the key range.

Below these are a set of string samples that also span the entire key range.

Whenever you play a key within this keyboard range, the sound produced will be a combination of the piano and the string sample.

In addition, in the example above, the user has arranged the piano samples into one group and the string samples in another. This is convenient since it allows for quick selection of the entire piano map, for example for balancing its level against the strings.

About Velocity Ranges

When zones are set up so that their key ranges overlap – completely or partially – you can use velocity switching and crossfading to determine which zones should be played back depending on how hard or soft you play on your MIDI keyboard.

This is done by setting up velocity ranges, with or without crossfading.

Each time you press a key on your MIDI keyboard, a velocity value between 1-127 is sent to Reason. If you press the key softly, a low velocity value is sent and if you press it hard, a high velocity value is sent.

This velocity value determines which samples will be played and which will not.

Let's say for example that you've mapped three different zones across the same key range:

→ Zone 1 has a velocity range from 1-40.

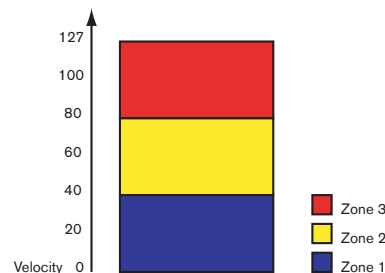
This means that the sample in it will be triggered by velocity values between 1-40.

→ Zone 2 has a velocity range of 41-80.

The sample in this zone will be played back by velocity values between 41-80.

→ Zone 3 has a velocity range of 81-127.

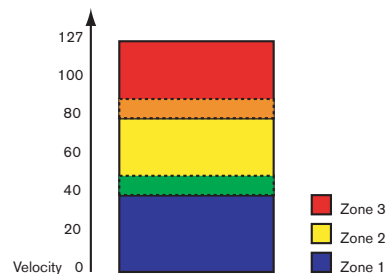
The sample in this zone will be triggered by all velocity values above 80.



Overlapping Velocity Ranges

Let's change the values above slightly:

- **Zone 1 has a velocity range from 1-60.**
- **Zone 2 has a velocity range of 41-100.**
- **Zone 3 has a velocity range of 81-127.**

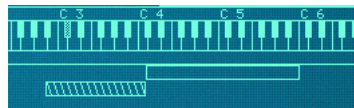


Now, velocity values between 41 and 60 will trigger samples from *both* Zone 1 and Zone 2. Likewise, velocity values between 81 and 100 will trigger sounds from Zone 2 and Zone 3.

About Full and Partial Velocity Ranges

You can see which zones have modified velocity ranges in the key map display:

- Zones with a full velocity range (0 - 127) are only shown with an outline.
- Zones with any other velocity range are shown as striped.



The top zone has a full velocity range (1-127), and the lower zone has a partial velocity range (any other range), which is indicated by stripes

Sorting Zones by Velocity Values

The Edit menu and the NN-XT context menu contain an item called "Sort Zones by Velocity". This option lets you automatically sort the selected zones in the display in descending order according to their set low or high velocity values.

When you invoke this option, the selected zones will be sorted from top to bottom starting with the one with the highest "Lo Vel" value.

Note however, that the sorting is done strictly on a *group* basis. That is, only zones that belong to the same group can be sorted in relation to each other.

If two zones have the same velocity range, they are sorted by key range.

Setting Velocity Range for a Zone

To set up a velocity range for a zone, proceed as follows:

1. **Select one or more zones that you want to adjust.**
2. **Use the knobs marked "Lo Vel" and "High Vel" in the sample parameter area to set the desired low- and high velocity values.**



Adjusting the "Lo Vel" value for a zone.

"Lo vel" is the lowest velocity value that should trigger the sample in the zone - i.e. if a key is pressed so softly that the velocity is lower than this value, the sample will not be played.

"Hi vel" is the highest velocity value that should trigger the sample, which means that if a key is pressed so hard that the velocity is higher than this value, the sample will not be played.

About Crossfading Between Zones

At the bottom right in the sample parameter area are two knobs marked "Fade In" and "Fade Out". These are primarily used for setting up velocity crossfades for smooth transitions between overlapping zones. In order to set up crossfades you adjust the fade out and fade in values for the overlapping zones.

Crossfading Between two Sounds

An example:

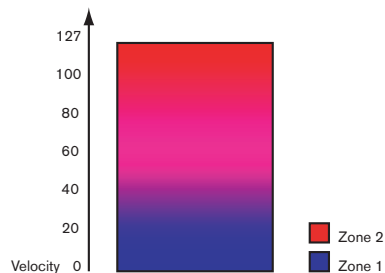
→ **Two zones are both set to play in the full velocity range of 1-127.**

→ **Zone 1 has a fade out value of 40.**

This means that this zone will play at full level with velocity values below 40, With higher velocity values, it will gradually fade out.

→ **Zone 2 has a fade in value of 80.**

This has the effect that as you play velocity values up to 80, this zone will gradually fade in. With velocity values above 80, it will play at full level.



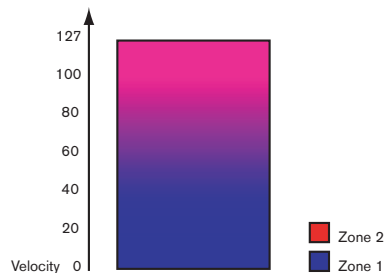
Another example:

Crossfading can be used to only fade in or fade out a certain sound. One common example is to set things up so that one sound plays the entire velocity range and another is faded in only at high velocity values.

→ **Zone 1 is set to play the entire velocity range with no crossfade.**

→ **Zone 2 is set to play the velocity range 80 to 127, with a fade in value of 110.**

This means that this zone will start fading in from velocity values 80 and will play at full level in the velocity range 110 to 127.



This can be used for example to add a rimshot to a regular snare sound or a harder attack to a softer violin sample.

Setting Crossfading for a Zone

Manually

To set up a crossfade for a zone, proceed as follows:

1. **Select one or more zones that you want to adjust.**
2. **Use the knobs marked “Fade In” and “Fade Out” in the sample parameter area, to set the desired values.**



- ★ You can change the values with finer precision by pressing [Shift] while turning the knobs, and you can reset the standard values by pressing [Command] (Mac)/[Ctrl] (Windows) and clicking on the knobs.

Automatically

If you find it tedious to manually set up crossfades between zones, NN-XT can do it for you! The Edit menu and the NN-XT context menu contain an item called “Create Velocity Crossfades”.

1. **Set up the zones so that their velocity ranges overlap, as desired.**
2. **Select the zones.**
You can select as many zones as you wish, not just one pair of overlapping zones.
3. **Select “Create Velocity Crossfades” from the Edit menu.**
NN-XT will analyze the overlapping zones and automatically set up what it deems to be appropriate fade in and fade out values for the zones.

Note the following important points:

- **This operation will not work if both zones have full velocity ranges.**
At least one of the zones must have a partial velocity range (see [page 170](#)).
- **This operation will not work if the zones are completely overlapping.**

Using Alternate

About the Alternate function



At the bottom right in the sample parameters area is a knob marked “Alt”. It only has two states - On and Off. This is used for semi-randomly alternating between zones during playback.

There are several practical uses for this. Here follows two examples:

- Layering several recordings of the same snare drum. By alternating between them you get a more natural repetition.
- Layering string up- and down strokes. By alternating you get the realistic effect of switching between the two directions of the stroke.

You can layer as many sounds as you will and the algorithm switches between them in a way that provides as little repetition as possible.

To set up an alternating set of zones, proceed as follows:

1. **Set up the zones so that they overlap completely or partially.**
2. **Select them all.**
3. **Set “Alt” to On for all the zones.**

Now, the program will automatically detect how to alternate between the zones, depending on their overlap.

Sample Parameters



The Sample parameter area is found below the screen. For details on how to adjust them (depending on whether one or more zones are selected) see [page 160](#). Below follows a run-down of the various parameters:

Root Note and Tune

These parameters are described on [page 167](#).

Sample Start and End

By turning the knobs you offset the start and end positions, so that they will play back more or less of a samples' waveform. Typical examples of use for this would be:

- ➔ **Removing unwanted portions from samples.**
This could be anything from noise to "dead air" at the beginning or end of a sample.
- ➔ **To create variations out of a single sample.**
These controls can be used to pick out any section of a recording for use as a sample.
- ➔ **Together with velocity sample start control.**
You can for example increase Sample Start and then apply negative velocity modulation to Sample Start. Then, the harder you play the more you will hear of the attack portion of the sound.

⚙️ If you hold down [Shift] when adjusting these parameters, the adjustment is always in single frames (samples).

Loop Start and End

A sample, unlike the cycles of an oscillator for example, is a finite quantity. There is a sample start and end. To get samples to play for as long as you press down the keys on your keyboard, they need to be *looped*.

For this to work properly, you have to first set up two loop points which determine the part of the sample that will be looped.

The instrument samples in the sound banks included with Reason are already looped. The same will be true for most commercial sample libraries. However, if you need to, you can use these controls to adjust the looping.

- The size and position of the loop – in the sample – is determined by two parameters, Loop Start (the beginning of the loop) and Loop End (the end point of the loop).
- The NN-XT then keeps repeating the section between the Loop Start and Loop end until the sound has decayed to silence.

Play Mode

By using this knob you can select one of the following loop modes for each zone:

- ➔ **FW**
The sample in the zone will play only once, without looping.
- ➔ **FW-LOOP**
The sample will play from the sample start point to the loop end point, jump back to the loop start point and then loop infinitely between the start and end loop points. This is the most common loop mode.
- ➔ **FW - BW**
The sample will play from the sample start point to the loop end point, then from the loop end point to the loop start point (backwards), and then loop infinitely forwards-backwards between the start and end loop points.
- ➔ **FW-SUS**
This works like FW-LOOP with the exception that it will only loop as long as the key is held down. As you release the key, the sample will play to the absolute end of the sample, that is beyond the boundaries of the loop.

This means that the sound may have a short natural release even if the release parameter is raised to a high value (which is not true for "FW-LOOP", where the release parameter always controls the length of the sound after the key is released).

- ➔ **BW**
The sample will play only once - from the end to the beginning - without looping.

Lo Key and Hi Key

These parameters are described on [page 164](#).

Lo Vel and Hi Vel

These parameters are described on [page 169](#).

Fade In and Fade Out

These parameters are described on [page 171](#).

Alt

This parameter is described on [page 172](#).

Out

The NN-XT features eight separate stereo output pairs (see [page 184](#)). For each zone, you can decide which of these output pairs to use. Thus, if you have created a key map consisting of eight zones, each of these can have a separate stereo output from NN-XT, and can then be routed to a separate mixer channel if you so wish.

- **To select which output a selected zone should be directed to, use the knob marked “Out” in the sample parameter area.**
The output pairs are indicated above the button.
- ! **Note that you still have to route the outputs the way you want them on NN-XT’s back panel. If you assign a zone to an output pair other than 1-2 (which is the default) no connections or auto routing are made. You have to do that manually.**

A Stereo Example

One possible way of utilizing this would be to create a drum kit. In this case you could load up to eight different stereo drum samples, assign them to separate outputs, route each to a separate mixer channel and then use the mixer to set levels and pan, add send effects etc.

Using a Stereo Output as Two Mono Outputs

If, on the other hand, you are using mono samples, you can use one stereo pair as a two separate outputs, effectively giving you a total of 16 separate outputs.

1. **Assign two zones to the same output.**
2. **Use the Pan control to pan one of the zones hard left and the other hard right.**
3. **Connect each of the two outputs in the stereo pair to a separate mixer channel.**

Group Parameters



The group parameters are located at the top left on the remote editor panel. These are parameters that in various ways are directly related to playing style.

Group parameters apply to a group, that is they are settings that are shared by all zones in a group.

- **To make adjustments to one group, select one or more zones that belong to the group, and adjust the parameter on the front panel.**
- **To set several groups to the same value, select at least one zone in each group you want to adjust, and adjust the parameter on the front panel.**

Key Poly

This setting determines the number of keys that you can play simultaneously (the polyphony). The maximum number is 99 and the minimum is 1, in which case the group will be monophonic.

Users of other samplers may want to note that the polyphony often means setting the number of *voices* that should be able to play. The NN-XT is different in this aspect, since the polyphony setting instead determines the number of *keys*, regardless of how many voices each key plays.

Legato and Retrig

Legato

Legato works best with monophonic sounds. Set Key Poly (see above) to 1 and try the following:

- **Hold down a key and then press another key without releasing the previous.**
Notice that the pitch changes, but the envelopes do not start over. That is, there will be no new “attack”.
- **If Key Poly is set to more voices than 1, Legato will only be applied when all the assigned keys are “used up”.**
For example, if you had a polyphony setting of “4” and you held down a 4 note chord, the next note you played would be Legato. Note, however, that this Legato key will “steal” one of the keys in the 4 note chord, as all the assigned keys were already used up!

Retrig

Retrig is the “normal” setting for playing polyphonic patches. That is, when you press a key without releasing the previous, the envelopes are triggered, like when you release all keys and then press a new one. In monophonic mode, Retrig has an additional function; if you press a key, hold it, press a new key and then release that, the first note is also retriggered.

LFO 1 Rate

This is used for controlling the rate of LFO 1 if it is used in “Group Rate” mode. In that case, this knob will take precedence over the rate parameter in the LFO 1 section. See [page 182](#) for detailed information about this.

Portamento

This is used for controlling portamento - a parameter that makes the pitch glide between the notes you play, rather than changing the pitch instantly as soon as you hit a key on your keyboard. By turning this knob you set how long it should take for the pitch to glide from one note to the next as you play them.

In legato mode, there will only be any portamento when actually playing legato (tied) notes.

With the knob turned all the way to the left, portamento is disabled.

Synth parameters

The Modulation controls



As previously described, the Modulation wheel (and the External Control wheel) can be used for controlling various parameters. These controls allow you to define which parameters the wheels should modulate and to what extent.

- **Below each of the knobs are the letters “W” and “X”.**
These are used for selecting the source that should control the parameter, and represent the “Modulation Wheel” and the “External Control wheel” respectively.
- **By clicking on any of the letters, you decide which source should control the parameter.**
You can select either, both or none. When a letter is “lit”, the corresponding source is set to control the parameter.
- **By turning the knobs, you decide how *much* the modulation and/or external control wheel should modulate the corresponding parameter.**

Note that all of the control knobs are bi-polar, which means that they can be set to both positive and negative values. Positive values are set by turning the knobs to the right, and negative values are thus set by turning the knobs to the left:

- Setting them to positive values means that the value of the controlled parameter will be raised if the source wheel is pushed forward.
- Setting them to negative values means that the value will be lowered when a wheel is pushed forward.
- Keeping the knobs in the center position means that no modulation control is applied.

There is one exception to these rules, and that is the LFO 1 Amt control, which works in a slightly different way. See below for more information about this.

The following parameters can be modulated:

F.Freq

This sets modulation control of the Filter's cutoff frequency (see [page 179](#)).

Mod Dec

This sets modulation control of the Decay parameter in the Modulation Envelope (see [page 180](#)).

LFO 1 Amt

This determines how much the amount of modulation from LFO 1 is affected by the Modulation wheel and/or the External Controller wheel. It does this by "scaling" the amounts set with the three destination knobs in the LFO 1 section (Pitch, Filter and Level, see [page 182](#)). We'll explain this with an example:

To use the Modulation Wheel to *increase* pitch modulation (vibrato), proceed as follows:

1. **Turn the Mod Wheel all the way down, so that no modulation is applied.**
2. **Activate the "W" button for LFO 1 Amt in the Modulation section.**
3. **Set the corresponding knob to "12 o'clock" (zero).**
4. **Set up LFO 1 so that as much vibrato is applied as you want it to be when the Modulation wheel is turned *all the way up*.**
5. **Increase LFO 1 Amt until you hear as much vibrato as you want it to be when the wheel is turned all the way down.**
If you turn LFO 1 Amt all the way up, there will be no vibrato at all when the wheel is all the way down.

To instead use the Modulation wheel to *decrease* vibrato, proceed as follows:

1. **Turn the Mod Wheel all the way down, so that no modulation is applied.**
2. **Activate the "W" button for LFO 1 Amt in the Modulation section.**
3. **Set the corresponding knob to "12 o'clock" (zero).**
4. **Set up LFO 1 so that as much vibrato is applied as you want it to be when the Modulation wheel is turned *all the way down*.**
5. **Turn the Modulation wheel all the way up.**
6. **Decrease LFO 1 Amt until you hear as much vibrato as you want it to be when the wheel is turned all the way up.**
If you turn LFO 1 Amt all the way down, there will be no vibrato at all when the wheel is all the way up.

F.Res

This sets modulation control of the Resonance parameter in the Filter (see [page 179](#)).

Level

This sets the amount of amplitude envelope modulation of each zone's level. The level set here will be the level of the highest point of the Amp Envelope.

LFO 1 Rate

This sets modulation control of the Rate parameter in LFO 1 (see [page 182](#)).

The Velocity controls



Velocity is used for controlling various parameters according to how hard or soft you play notes on your keyboard. A typical use of velocity control is to make sounds brighter and louder if you strike a key harder. By using the knobs in this section, you can control if and how much the various parameters will be affected by velocity.

Just like the modulation controls, all of the velocity control knobs are bi-polar, and can be set to both positive and negative values.

- Setting them to positive values means that the value of the controlled parameter will be raised the harder you play.
- Setting them to negative values means that the value will be lowered the harder you play.
- Keeping the knobs in the center position means that no velocity control is applied.

The following parameters can be velocity controlled:

F.Freq

This sets velocity control of the Filter's cutoff frequency (see [page 179](#)).

Mod Dec

This sets velocity control of the Decay parameter in the Modulation Envelope (see [page 180](#)).

Level

This sets velocity control of the Amp Envelope.

Amp Env Attack

This sets velocity control of the Attack parameter in the Amplitude Envelope (see [page 181](#)).

S. Start

This sets velocity control of the Sample Start parameter (see [page 173](#)), so that it will be offset forwards or backwards, according to how hard or soft you play.

This allows you to control how much of the attack portion of the sample you hear when playing harder or softer.

To be able to make use of negative values for this parameter, you must increase the sample parameter Sample Start.

The Pitch section



This section contains various parameters related to controlling the pitch, or frequency, of the zones.

Pitch Bend Range

This lets you set the amount of pitch bend, i.e. how much the pitch changes when you turn the pitch bend wheel fully up or down. The maximum range is +/- 24 semitones (2 Octaves).

Setting the pitch

Use the three knobs marked "Octave", "Semi" and "Fine" to change the pitch of the sample(s):

→ **Octave**

This changes the pitch in steps of one full octave. The range is -5 – 0 – 5.

→ **Semi**

This lets you change the pitch in semitone steps. The range is -12 – 0 – 12 (2 octaves).

→ **Fine**

This changes the pitch in cents (hundredths of a semitone). The range is -50 – 0 – 50 (down or up half a semitone).

K. Track

This knob controls Keyboard Tracking of the pitch.

- In the center position, each key represents a semitone This is the normal setting.
- When turned all the way down, all keys play the same pitch. This can be useful for percussion like timpani where you might want to play the same pitch from a range of keys.
- When turned all the way up, each key on the keyboard shifts the pitch one octave.

The Filter Section



Filters can be used for shaping the character of the sound. The filter in NN-XT is a multimode filter with six different filter types.

- **To activate/deactivate the filter, click the On/Off button in the top right corner.**

When the filter is activated, the button is lit.

Filter mode

To select a filter mode, either click the Mode button in the bottom right corner or click directly on the desired filter name so that it lights up:

- **Notch**

The notch filter is used for cutting off frequencies in a narrow frequency range around the set cutoff frequency, while letting the frequencies below and above through.

- **HP 12**

This is a highpass filter with a 12 dB/Octave roll-off slope. A highpass filter cuts off low frequencies and lets high frequencies pass. That is, frequencies below the cutoff frequency are cut off and frequencies above it pass through.

- **BP 12**

This is a bandpass filter with a 12 dB/Octave roll-off slope. A bandpass filter could be viewed as the opposite of a notch filter. It cuts off both the high and the low frequencies, while frequencies in the band range pass through.

- **LP 6**

This is a lowpass filter with a gentle, 6 dB/Octave slope. A lowpass filter is the opposite of a highpass filter. It lets the low frequencies through and filters out the high frequencies. This filter has no Resonance.

- **LP 12**

This is a lowpass filter with a 12 dB/Octave roll-off slope.

- **LP 24**

This is a lowpass filter with a fairly steep roll-off slope of 24 dB/Octave.

Filter controls

The following filter controls are available:

- **Freq**

This is used for setting the filter cutoff frequency. The cutoff frequency determines the limit above or below which frequencies will be cut off depending on the selected filter type. In the case of a lowpass filter for example, frequencies below the cutoff frequency will be allowed to pass through, while frequencies above it will be cut off. The farther to the right you turn the knob, the higher the cutoff frequency will be.

- ★ **It is very common to modulate filter frequency with the modulation envelope, as described on [page 180](#).**

- **Res**

Technically, this knob controls feedback of the output signal from the filter, back to its input. Acoustically it emphasizes frequencies around the cutoff frequency. For a lowpass filter for example, increasing Res will make the sound increasingly more hollow until the sound starts "ringing". If you set a high value for the Res parameter and then vary the filter frequency, this will produce a classic synthesizer filter sweep.

For the notch and bandpass filter types, the Resonance setting instead adjusts the width of the band. That is, the higher the resonance setting, the narrower the band will be where frequencies are cut off (notch) or let through (Bandpass).

- **K. Track**

This lets you activate and control keyboard tracking of the filter frequency. If keyboard tracking is activated, the set cutoff frequency of the filter will change according to the notes you play on your keyboard. That is, if you play notes higher up on the keyboard, the filter frequency will be raised and vice versa.

When the knob is set to its center position, filter frequency is adjusted so that the harmonic content remains constant across the keyboard.

Keyboard tracking is deactivated by default (the knob all the way to the left). This means that the filter frequency will remain unchanged regardless of where on the keyboard you play.

The Modulation Envelope



The Modulation Envelope parameters let you control how certain parameters, or destinations, should change over time - from the moment a note is struck to the moment it is released again.

The destinations you can use are:

- Pitch
- Filter frequency

Parameters

The following are the available controlling parameters:

→ Attack

When you press a key on your keyboard, the envelope is triggered. The attack parameter then controls how long it should take before the controlled parameter (pitch or filter) reaches the maximum value, when you press a key. By setting attack to a value of "0", the destination parameter would reach the maximum value instantly. By raising the attack parameter, the value will instead slowly "slide" up to its maximum.

→ Hold

This is used for deciding how long the controlled parameter should stay at its maximum value before starting to decrease again. This can be used in combination with the Attack and Decay parameters to make a value reach its maximum level, stay there for a while (hold) and then start dropping gradually down to the sustain level.

→ Decay

After the maximum value for a destination has been reached and the Hold time has expired, the controlled parameter will start to gradually drop down to the sustain level. How long it should take before it reaches the sustain level is controlled with the Decay parameter. If Decay is set to "0", the value will immediately drop down to the sustain level.

→ Sustain

The Sustain parameter determines the value the envelope should drop back to after the Decay. If you set Sustain to full level however, the Decay setting doesn't matter since the value will never decrease.

A combination of Decay and Sustain can be used for creating envelopes that rise up to the maximum value, then gradually decrease to, and stay on a level somewhere in-between zero and maximum.

→ Release

This works just like the Decay parameter, with the exception that it determines the time it takes for the value to fall back to zero *after* the key is released.

→ Delay

This is used for setting a delay between when a note is played and when the effect of the envelope starts. That is, the sound will start unmodulated, and the envelope will kick in after you have kept the key(s) pressed down for a while. Turn the knob to the right to increase the delay time. If the knob is set all the way to the left, there will be no delay.

→ Key To Decay

By using this, you can cause the value of the Decay parameter (see above) to be offset depending on where on your keyboard you play. If you turn the knob to the right the decay value will be raised the higher up you play, and turning the knob to the left will lower the decay value the higher up you play. With the knob in the center position, this parameter is deactivated.

Destinations

The following are the available Mod Envelope destinations:

→ Pitch

This will make the envelope modulate the pitch, as set in the Pitch section (see [page 178](#)). Turn the knob to the right to raise the pitch and to the left to lower the pitch. In the middle position, pitch will not be affected by the envelope.

→ Filter

This will make the envelope modulate the cutoff frequency of the Filter (see [page 179](#)). Turn the knob to the right to increase the frequency and to the left to lower the frequency. In the middle position, the envelope will have no effect on the cutoff frequency.

The Amplitude Envelope



The Amplitude Envelope parameters let you control how the volume of a sound should change over time - from the moment a note is struck to the moment it is released again.

Parameters

Most of the Amplitude Envelope parameters are identical to those of the Modulation Envelope. So for a detailed description of the following parameters, please refer to the modulation envelope section on [page 180](#):

- Attack
- Hold
- Decay
- Sustain
- Release
- Delay
- Key To Decay

The following are the parameters that are unique for the Amp Envelope section:

→ Level

This knob sets the level of the zone. Turn it to the right to raise the level.

→ Spread and Pan modes

These two parameters are used for controlling the stereo (pan) position of the sound. The Spread knob determines the sound's width in the stereo image (how far left – right the notes will be spread out). If this is set to "0", no spread will take place. The Mode selector switch is used for choosing which type of spread you want to apply:

Mode	Description
Key	This will make the pan position shift gradually from left to right, the higher up on the keyboard you play.
Key 2	This will make the pan position shift from left to right and then back again from right to left in a sequence of eight keys. Playing 4 adjacent semitones thus makes the pan position gradually go from left to right. The next 4 higher semitone notes will then change the pan position from right to left in the same way, and this cycle will then be repeated.
Jump	This will make the pan position jump between left and right each time a note is played.

→ Pan

This controls the stereo balance of the output pair to which a zone is routed. In the middle position, the signal appears equally strong on the left and right channel in a stereo pair. By turning the knob to the left or right, you can change the stereo balance.

Note that if you for instance turn the Pan knob all the way to the left, you cause the signal to be output from the left channel of the stereo pair *only*.

You can use this to treat a stereo output as two independent mono outputs, if required.

See [page 174](#) for information on routing zones to output pairs.

The LFOs



NN-XT features two Low Frequency Oscillators - LFO 1 and LFO 2. “Normal” oscillators generate a waveform and a frequency, and produce sound. *Low frequency* Oscillators on the other hand, also generate a waveform and a frequency, but there are two major differences:

- LFOs only generate sounds of a low frequency.
- LFOs don’t produce sound, but are instead used for modulating various parameters.

The most typical use of an LFO is to modulate the pitch of a sound (generated by an oscillator or - in the case of NN-XT - a sample), to produce vibrato.

About the Difference between LFO 1 and LFO 2

There are two fundamental differences between LFO 1 and LFO 2:

- **LFO 2 is always key synced, that is, each time you press a key, the LFO waveform starts over from scratch. LFO 1 can be switched between key synced and non-key synced modes.**
- **LFO 2 only has one waveform, triangle.**

The following parameters are available for the LFOs:

Rate (LFO 1 and 2)

This knob controls the frequency of the LFO. For a faster modulation rate, turn the knob to the right.

The Rate knob of LFO 1 is also used for setting the timedivision when synchronizing the LFO to the song tempo (see below).

Delay (LFO 1 and 2)

This can be used for setting a delay between when a note is played and when the LFO modulation starts kicking in (gradually). This way, you can make the sound start unmodulated, and then have the LFO modulation start after you have kept the key(s) pressed down for a while.

Turn the knob to the right to increase the delay time.

Mode (LFO 1 only)

This lets you set the “operation mode” for the LFO. Click the button to switch between the available modes:

→ Group Rate

In this mode, the LFO will run at the rate set for it’s group in the group section, rather than at the rate set here (see [page 174](#)). This way, all zones in the group will have the exact same modulation rate.

→ Tempo Sync

In this mode, the LFO will be synchronized to the song tempo, in one of 16 possible time divisions.

! When tempo sync is activated, the Rate knob is used for selecting the desired timedivision. Turn the Rate knob and observe the tool tip for an indication of the timedivision.







→ Free Run

In free run mode, the LFO simply runs at the rate set with the Rate parameter. Furthermore, if Key Sync is deactivated, the modulation cycle will not be retrigged each time you press a key - it will run continuously.

Waveform (LFO 1 only)

Here, you select which type of waveform should be used for modulating the destination parameters.

Click the button to switch between the following waveforms:

Waveform	Description
Triangle 	This is a smooth waveform, suitable for normal vibrato.
Inverted Sawtooth 	This produces a "ramp up" cycle. If applied to an oscillator's frequency, the pitch would sweep up, after which the cycle immediately starts over.
Sawtooth 	This produces a "ramp down" cycle, the same as above but inverted.
Square 	This produces cycles that abruptly change between two values, usable for trills etc.
Random 	Produces random stepped modulation to the destination. Some vintage analog synths called this feature "sample & hold".
Soft Random 	The same as above, but with smooth modulation.

! **LFO 2 always uses a triangle waveform.**

Key Sync (LFO 1 only)

By activating key sync, you "force" the LFO to restart its modulation cycle each time a key is pressed.

! **Note that LFO 2 always uses Key Sync.**

Destinations for LFO 1

The following parameters can be modulated by LFO 1:

→ Pitch

This will make the LFO modulate the pitch, for vibrato, trills, etc. It can be set to -2400 – 0 – 2400 cents which equals 4 octaves. The set pitch will change up and down by this amount, with each modulation cycle. Turning the knob to the right will make the modulation cycle start above the set pitch, while turning it to the left will invert the cycle. Keeping this in the middle position means that the pitch will not be affected by the LFO.

→ Filter

This will make the LFO modulate the cutoff frequency of the Filter, for auto-wah effects, etc. The positive/negative effect is the same as for pitch.

→ Level

This will make the LFO modulate NN-XT's output level, for tremolo effects, etc. The positive/negative effect is the same as for pitch.

Destinations for LFO 2

The following parameters can be modulated by LFO 2:

→ Pan

This makes the LFO modulate the pan position of a zone. The sound will move back and forth in the stereo field. Turning the knob to the left makes the sound move from left to right, and turning it to the right thus makes it move from right to left. The middle position provides no modulation at all.

→ Pitch

Just like for LFO 1 (see above), this makes LFO 2 modulate the pitch. The range is also the same as for LFO 1.

Connections

On the back panel of NN-XT are a number of connectors. Many of these are CV/Gate related. Using CV/Gate is described in the chapter “Routing Audio and CV”.



Sequencer Control

The Sequencer Control CV and Gate inputs allow you to play the NN-XT from another CV/Gate device (typically a Matrix or a Redrum). The signal to the CV input controls the note pitch, while the signal to the Gate input delivers note on/off along with velocity.

Modulation Input

These control voltage (CV) inputs (with associated voltage trim pots), can modulate various NN-XT parameters from other devices. These inputs can control the following parameters:

- Oscillator Pitch
- Filter Cutoff Frequency
- Filter Resonance
- LFO 1 Rate
- Master Volume
- Pan
- Modulation Wheel

Gate Input

These inputs can receive a CV signal to trigger the following envelopes:

- Amplitude Envelope
- Modulation Envelope

Note that connecting to these inputs will override the normal triggering of the envelopes. For example, if you connect a Matrix Gate Out to the Gate In Amp Envelope, you would not trigger the amp envelope by playing notes, as this is now controlled by the Matrix Gate Out. In addition you would only hear the Gate Out triggering the envelope for the notes that you hold down.

Audio Output

There are 16 audio output jacks on the NN-XT's back panel - eight separate *stereo pairs*. When you create a new NN-XT device, the first output pair (1L & 2R) is auto-routed to the first available channel on the audio mixer.

The other output pairs are never automatically routed. If you wish to use any of the other output pairs, you have to manually connect them to the desired device - typically a mixer channel. The basics on Routing is described in the chapter “Managing the Rack” in the Getting Started book.

! Note that when you use any other output pair than the first, you also have to route one or more zones to it if you want it to actually output sound, since all zones by default are routed to outputs 1 & 2. How to route zones to other outputs is described on [page 174](#).



REASON

16

→ Dr. Rex Loop Player

Adding a Loop

To add a loop into the Dr.Rex Loop Player, proceed as follows:

1. **Open the browser by selecting “Browse ReCycle/REX Files” from the Edit menu or the device context menu, or click on the folder button beside the Loop name display.**



2. **In the browser, locate and open the desired loop.**
You can listen to the loops before loading by using the Preview function in the browser.

! **Loading a new REX file will replace any currently loaded file.**

Auditioning the Loop in Dr.Rex

- **Once loaded, you can check out the loop by using the Preview button.**

It will play back repeatedly in the tempo set on the transport panel. If you change the tempo, the loop tempo follows.



- **You can also play the loop once via MIDI, by using the D0 key.**
- **To check out the loop together with other device sequencer data and patterns already recorded, activate both the Preview function and the sequencer Play button.**

This does not have to be done in any particular order, they will play back in perfect sync anyway.

Loading Loops “On the Fly”

Another practical method for checking out loops, is to load them “on the fly”, i.e. during playback. This is especially useful if you want to check out a number of loops against other sequencer data and patterns previously recorded. Proceed as follows:

1. **Activate Preview on the Dr.Rex and start sequencer playback.**
The REX loop and the sequencer are synced.
 2. **Now load a new REX file by using the Browser in one of the usual ways.**
After a brief silence, the new file is loaded, and sync is maintained.
 3. **Repeat step 2 as necessary until you have found a suitable loop.**
- **If you are trying out loops within the same folder, the quickest ways to select a new loop is to use the arrow keys next to the loop name display.**
Or, you can click in the loop name display and select a new loop from the pop-up menu that appears.
- ! **Note that the Preview function is not the “real” way of playing back REX loops. If you want to use the loop in a context with other devices, you should transfer the REX slices to notes in the sequencer, as described on [page 188](#).**

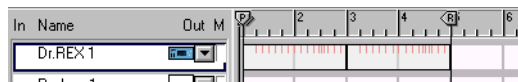
Creating Sequencer Notes

To be able to make your REX loop start at the same time as other sequencer or pattern data, you first have to create sequencer notes from the slices:

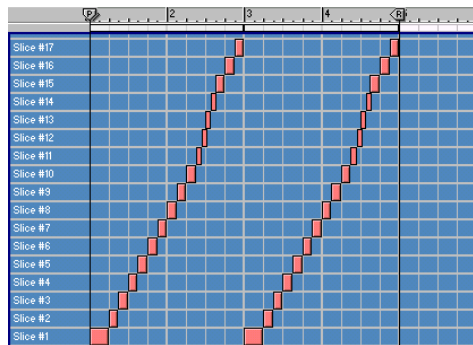
1. **Select a sequencer track connected to the Dr.Rex device.**
2. **Set the left and right locators to encompass the section you want to fill with REX notes.**
You may want to make sure that this area doesn't contain any notes already, to avoid confusion.
3. **Click the To Track button on the Dr.Rex panel.**



Now, the program will create a note for each slice, positioned according to the timing of the slices. The notes will be pitched in semitone steps, with the first note on C#1, the second on C#1 and so on, with one pitch for each slice. If the area between the locators is longer than the loop length, the loop notes will be repeated to fill out the loop.



The loop notes in the Arrange View...



...and in the Edit View.

Activating playback in the sequencer will now play back the notes on the sequencer track. These in turn will play back the slices in the Dr.Rex device, in the correct order and with the original timing maintained. Now the fun begins!

- You can change the groove in the loop by quantizing or moving notes.
- You can transpose notes to change the order of the slices on playback.
- You can use the Alter Notes function in the Change Events dialog (see [page 32](#)) to scramble the loop notes - without destroying the original loop timing.
- You can remove and draw new notes, creating any kind of variation.
- You can use the User Groove function to apply the rhythmic feel of the loop to notes on other sequencer tracks.

For details about editing in the sequencer, see [page 22](#).

! Note that if you have created sequencer notes from a REX file, you cannot load a new REX file into Dr.Rex and play it from the existing track. Well, you can, but it will not play back properly. If you have created notes in this way, and want to change the REX file, first delete the notes, then use the “To Track” command again *after* having loaded the new REX file.

You can also export the REX file as a MIDI file, as described on [page 258](#).

Slice Handling

Selecting Slices



A selected slice is indicated by being highlighted in the waveform display. To select a slice, use one of the following methods:

- **By clicking in the waveform display.**
If you hold down [Option] (Mac) or [Alt] (Windows) and click on a slice in the waveform display, it will be played back. The pointer takes on the shape of a speaker symbol to indicate this.
- **By using the “Slice” knob below the waveform display.**
- **Via MIDI.**
If you activate “Select Slice Via MIDI”, you can select and “play” slices using your MIDI keyboard. Slices are always mapped to consecutive semitone steps, with the first slice always being on the “C1” key.



- **If you play back a loop with “Select via MIDI” option activated, each consecutive slice is selected as it is played back.**
You can edit parameters during playback.

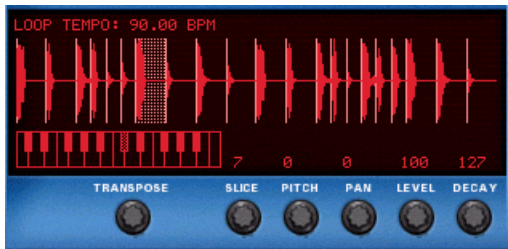
Editing Individual Slices

There are two basic methods you can use to edit individual slices in Reason:

- **In the Waveform display of the Dr.Rex device.**
This is used for making playback settings for a slice.

- **In the Sequencer.**
Here you can edit the notes that play the slices. There is a special REX lane for editing REX slice notes, with the notes indicated by the slice numbers instead of by pitches. Editing in the sequencer is described in the Sequencer chapter.

Editing in the Waveform Display



Here you are able to edit several parameters for each slice, by first selecting it, and then using the knobs below the waveform display. The following slice parameters can be set:

Parameter	Description
Pitch	Allows you to transpose each individual slice in semitone steps, over a range of more than eight octaves.
Pan	The stereo position of each slice.
Level	The volume of each slice. The default level is 100.
Decay	Allows you to shorten individual slices.

- ! **If you have made settings to any of the parameters listed above, these will be lost if you load a new REX file. All Dr.Rex panel settings are stored in the Song. You cannot directly apply panel settings to another REX file!**

Dr.Rex Synth Parameters

The Dr.Rex synth parameters are used for shaping and modulating the sound of the REX loops. These parameters are familiar synth parameters, similar to the ones in the synthesizers; The Subtractor and the Malström, and in the samplers; the NN-19 and the NN-XT. It is important to remember that these parameters do not alter the REX files in any way, only the way they will play back.

! **These parameters are global, in the sense that they will affect all slices in a REX file.**

Oscillator Section



For a REX file, the audio contained in the slices are what oscillators are for a synthesizer, the main sound *source*. The following settings can be made in the Osc section of the Dr.Rex:

Setting the overall Pitch

You can change the pitch of a REX file in three ways:

→ **In octave steps.**

This is done using the Oct knob. The range is 0 - 8, with "4" the default.

→ **In semitone steps.**

This is done by using the Transpose knob below the waveform display, or by clicking on the keyboard above the knob. You can raise or lower the frequency in 12 semitone steps (+/- 1 octave). The transpose value can also be changed via MIDI, by pressing a key between C-2 and C0 (with C1 resetting the transpose value to zero).

→ **In cents (hundredths of a semitone).**

The range is -50 to 50 (down or up half a semitone).

! **To tune an individual slice, you select it and use the Pitch parameter below the waveform display.**

Osc Envelope Amount

This parameter determines to what degree the overall pitch of the REX file will be affected by the Filter Envelope (see [page 191](#)). You can set negative or positive values here, which determines whether the envelope curve should raise or lower the pitch.

The Filter Section



Filters are used for shaping the overall timbre of the sound. The filter in Dr.Rex is a multimode filter with five filter modes.

→ **You activate or deactivate the filter completely by clicking the Filter button.**

The filter is active when the button is lit.

Filter Mode

With this selector you can set the filter to operate as one of five different types of filter. These are as follows:

→ **24 dB Lowpass (LP 24)**

Lowpass filters let low frequencies pass and cut out the high frequencies. This filter type has a fairly steep roll-off curve (24dB/Octave). Many classic synthesizers (Minimoog/Prophet 5 etc.) used this filter type.

→ **12 dB Lowpass (LP 12)**

This type of lowpass filter is also widely used in classic analog synthesizers (Oberheim, early Korg synths, etc.). It has a gentler slope (12 dB/Octave), leaving more of the harmonics in the filtered sound compared to the LP 24 filter.

→ **Bandpass (BP 12)**

A bandpass filter cuts both high and low frequencies, while midrange frequencies are not affected. Each slope in this filter type has a 12 dB/Octave roll-off.

→ **High-Pass (HP12)**

A highpass filter is the opposite of a lowpass filter, cutting out the lower frequencies and letting the high frequencies pass. The HP filter slope has a 12 dB/Octave roll-off.

→ **Notch**

A notch filter (or band reject filter) could be described as the opposite of a bandpass filter. It cuts off frequencies in a narrow midrange band, letting the frequencies below and above through.

Filter Frequency

The Filter Frequency parameter (often referred to as “cutoff”) determines which area of the frequency spectrum the filter will operate in. For a lowpass filter, the frequency parameter could be described as governing the “opening” and “closing” of the filter. If the Filter Freq is set to zero, none or only the very lowest frequencies are heard, if set to maximum, all frequencies in the waveform are heard. Gradually changing the Filter Frequency produces the classic synthesizer filter “sweep” sound.

! **Note that the Filter Frequency parameter is usually controlled by the Filter Envelope (see [page 191](#)) as well. Changing the Filter Frequency with the Freq slider may therefore not produce the expected result.**

Resonance

The filter resonance parameter affects the character of the filter sound. For lowpass filters, raising the resonance will emphasize the frequencies around the set filter frequency. This produces a generally thinner sound, but with a sharper, more pronounced filter frequency “sweep”. The higher the resonance value, the more resonant the sound becomes until it produces a whistling or ringing sound. If you set a high value for the resonance parameter and then vary the filter frequency, this will produce a very distinct sweep, with the ringing sound being very evident at certain frequencies.

- For the highpass filter, the resonance parameter operates just like for the lowpass filters.
- When you use the Bandpass or Notch filter, the resonance setting adjusts the width of the band.
When you raise the resonance, the band where frequencies are let through (Bandpass), or cut (Notch) will become narrower. Generally, the Notch filter produces more musical results using low resonance settings.

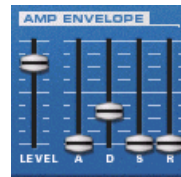
Envelope Section

Envelope generators are used to control several important sound parameters in analog synthesizers, such as pitch, volume, filter frequency etc. In a conventional synthesizer, envelopes govern how these parameters should respond over time - from the moment a note is struck to the moment it is released. In the Dr.Rex device however, the envelopes are triggered each time a slice is played back.

There are two envelope generators in the Dr.Rex, one for volume, and one for the filter frequency (and/or pitch). Both have the standard four parameters; Attack, Decay, Sustain and Release

! **Please refer to the Subtractor chapter for a description of the basic envelope parameters.**

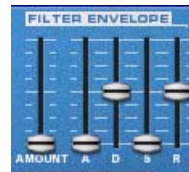
Amplitude Envelope



The Amp Envelope governs how the volume of a slice should change over time, from the time it is triggered (the slice note starts) until the slice note ends. This can be used to make a loop more distinct (by having a snappy attack and a short decay time) or more spaced-out (by raising the attack time).

The Level parameter acts as a general volume control for the loop.

Filter Envelope



The Filter Envelope can be used to control two parameters; filter frequency and overall loop pitch. By setting up a filter envelope you control how the filter frequency and/or the pitch should change over time for each slice.

The Amount parameter determines to what degree the filter frequency will be affected by the Filter Envelope. The higher the Amount setting, the more pronounced the effect of the envelope on the filter.

🔗 **Try lowering the Frequency slider and raising Resonance and Envelope Amount to get the most effect of the filter envelope!**

LFO Section



LFO stands for Low Frequency Oscillator. LFOs are oscillators in the sense that they generate a waveform and a frequency. However, there are two significant differences compared to normal sound generating oscillators:

- LFOs only generate waveforms with low frequencies.
- The output of the two LFOs are never actually heard. Instead they are used for modulating various parameters.

The most typical application of an LFO is to modulate the pitch of a (sound generating) oscillator or sample, to produce vibrato. In the Dr.Rex device, you can also use the LFO to modulate the filter frequency or panning.

Waveform

LFO 1 allows you to select different waveforms for modulating parameters. These are, from top to bottom:

Waveform	Description
Triangle	This is a smooth waveform, suitable for normal vibrato.
Inverted Sawtooth	This produces a “ramp up” cycle. If set to control pitch (frequency), the pitch would sweep up to a set point (governed by the Amount setting), after which the cycle immediately starts over.
Sawtooth	This produces a “ramp down” cycle, the same as above but inverted.
Square	This produces cycles that abruptly changes between two values, usable for trills etc.
Random	Produces random stepped modulation to the destination. Some vintage analog synths called this feature “sample & hold”.
Soft Random	The same as above, but with smooth modulation.

Destination

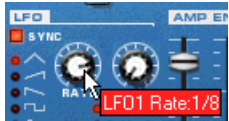
The available LFO Destinations are as follows:

Destination	Description
Osc	Selecting this makes LFO control the pitch (frequency) of the REX file.
Filter	Selecting this makes the LFO control the filter frequency.
Pan	Selecting this makes the LFO modulate the pan position of the REX file, i.e. it will move the sound from left to right in the stereo field.

Sync

By clicking this button you activate/deactivate LFO sync. The frequency of the LFO will then be synchronized to the song tempo, in one of 16 possible time divisions. When sync is activated, the Rate knob (see below) is used for setting the desired time division.

Turn the knob and check the tooltip for an indication of the time division.



Rate

The Rate knob controls the LFO's frequency. Turn clockwise for a faster modulation rate.

Amount

This parameter determines to what degree the selected parameter destination will be affected by the LFO 1, i.e. the amount of vibrato, filter wah or auto-panning.

Velocity Control



Velocity is usually used to control various parameters according to how hard or soft you play notes on your keyboard. A REX file does not contain velocity values on its own. And when you create sequencer track data by applying the “To Track” function, all velocities are set to a default value of “64”. As velocity information is meant to reflect *variation*, having them all set to the same value is not meaningful if you wish to velocity control Dr.Rex parameters.

There are basically two ways you can apply “meaningful” velocity values to REX files:

- After creating track data, you can edit velocity values in the Velocity Lane in the sequencer.
- You can play slices in real time on your keyboard. The resulting data will have velocity values reflecting how the notes were struck when you played.

When velocity values have been adjusted, you can control how much the various parameters will be affected by velocity. The velocity sensitivity amount can be set to either positive or negative values, with the center position representing no velocity control.

The following parameters can be velocity controlled:

Parameter	Description
Amp	This let's you velocity control the overall volume of the file. If a positive value is set, the volume will increase with higher velocity values.
F. Env	This sets velocity control for the Filter Envelope Amount parameter. A positive value will increase the envelope amount with higher velocity values. Negative values invert this relationship.
F. Decay	This sets velocity control for the Filter Envelope Decay parameter. A positive value will increase the Decay time with higher velocity values. Negative values invert this relationship.

Pitch Bend and Modulation Wheels



The Pitch Bend wheel is used for “bending” the pitch up or down. The Modulation wheel can be used to apply various modulation while you are playing the loop. Virtually all MIDI keyboards have Pitch Bend and Modulation controls. Dr.Rex also has two functional wheels that could be used to apply real time modulation and pitch bend should you not have these controllers on your keyboard, or if you aren’t using a keyboard at all. The wheels mirror the movements of the corresponding MIDI keyboard controllers.

Pitch Bend Range

The Range parameter sets the amount of pitch bend when the wheel is turned fully up or down. The maximum range is “24” (=up/down 2 Octaves).

Modulation Wheel

The Modulation wheel can be set to simultaneously control a number of parameters. You can set positive or negative values, just like in the Velocity Control section. The following parameters can be affected by the modulation wheel:

Parameter	Description
F. Freq	This sets modulation wheel control of the filter frequency parameter. A positive value will raise the frequency if the wheel is pushed forward. Negative values invert this relationship.
F. Res	This sets modulation wheel control of the filter resonance parameter. A positive value will increase the resonance if the wheel is pushed forward. Negative values invert this relationship.
F. Decay	This sets modulation wheel control for the Filter Envelope Decay parameter. A positive value will increase the decay if the wheel is pushed forward. Negative values invert this relationship.

Setting Number of Voices - Polyphony



This determines the polyphony, i.e. the number of voices, or slices, Dr. Rex can play simultaneously. For normal loop playback, it is worth noting that slices sometimes “overlap”. Therefore, it is recommended that you use a polyphony setting of about 3-4 voices when playing REX files. If you are “playing” slices via MIDI, the polyphony setting should be set according to how many overlapping slices you want to have.

! Note that the Polyphony setting does not “hog” voices. For example, if you are playing a file that has a polyphony setting of ten voices, but the file only uses four voices, this won’t mean that you are “wasting” six voices. In other words, the polyphony setting is not something you need to consider if you want to conserve CPU power - it is only the number of voices *actually used* that counts.

Audio Quality Settings



These two parameters provide ways of balancing audio quality vs. conservation of computer power.

High Quality Interpolation

When this is activated, the loop file playback is calculated using a more advanced interpolation algorithm. This results in better audio quality, especially for loops with a lot of high frequency content.

→ **High Quality Interpolation uses more computer power - if you don't need it, it's a good idea to turn it off!**

Listen to the loop in a context and determine whether you think this setting makes any difference.

! If you are using a Macintosh with a G4 (AltiVec) processor, turning High Quality Interpolation off makes no difference.

Low Bandwidth (BW)

This will remove some high frequency content from the sound, but often this is not noticeable (this is especially true if you have “filtered down” your loop). Activating this mode will save you some extra computer power, if needed.

Connections

On the back panel of Dr.Rex you will find the connectors, which are mostly CV/Gate related. Using CV/Gate is described in the chapter “Routing Audio and CV”.

Audio Outputs

These are the main left and right audio outputs. When you create a new Dr.Rex device, these are auto-routed to the first available channel on the audio mixer.

Slice Gate Output

This outputs a gate signal for each triggered slice in the loop.

Modulation Inputs

These control voltage (CV) inputs (with associated voltage trim pots), allow you to modulate various Dr.Rex parameters from other devices (or from the modulation outputs of the Dr.Rex device itself). The following CV inputs are available:

- Osc Pitch.
- Filter Cutoff.
- Filter Resonance.
- Amp Level.
- Mod Wheel.

Modulation Outputs

The Modulation outputs can be used to voltage control other devices, or other parameters in the Dr.Rex device itself. The Modulation Outputs are:

- Filter Envelope.
- LFO.

Gate Inputs

These inputs can receive a CV/gate signal to trigger the two envelopes. Note that connecting to these inputs will override the “normal” triggering of the envelopes. For example, if you connected an LFO CV output on another device to the Gate Amp input on the Dr.Rex, the amplitude envelope would not be triggered by the incoming MIDI notes to the Dr.Rex device, but by the LFO CV signal. In addition you would only hear the LFO triggering the envelope for the slices that were playing at the moment of the trigger.

- Amp Envelope
- Filter Envelope



REASON

17

→ Matrix Pattern Sequencer

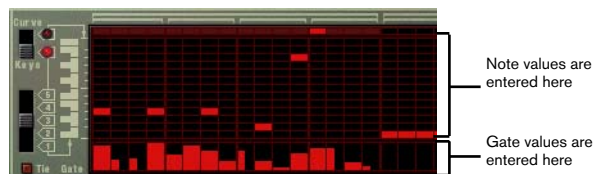
Introduction



The Matrix is a pattern-based device. Matrix doesn't generate sound on its own, but has to be connected to another instrument device. It basically works by sending pattern data in the form of Note CV (pitch) and Gate CV (note on/off plus velocity) or Curve CV (for general CV parameter control) signals to a device or device parameter. The patterns can be up to 32 steps, and there are 32 memory locations for storing pattern data. The Matrix is monophonic and can control one voice in an instrument device.

Unlike most other devices in Reason, the user interface of the Matrix is not modeled on any existing hardware equivalent. The hardware devices that could be said to have similar functionality are analog step sequencers, which usually had rows of knobs that controlled the note pitch and gate values for each step.

About the Three Output Types



Note and Gate CV values.

The Matrix can produce three types of output: Curve CV, Note (Key) CV and Gate CV.

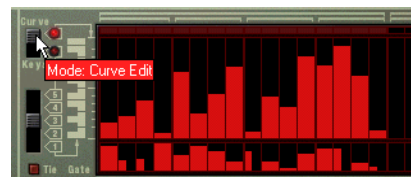
→ Note CV normally controls note pitch.

When connected to an instrument device Sequencer Control input, the values correspond to semitone steps.

→ Gate CV represents a note-on/off value, plus a *level* value (that could be likened to velocity).

Both of these two outputs are typically connected to the Sequencer Control Gate and CV inputs on a compatible instrument device. For example, if you create a Matrix with either a synthesizer (Subtractor, Malström) or a sampler (NN-19, NN-XT) selected, they will be auto-routed in this way, and will control one voice in the device.

→ Curve CV is a separate pattern, programmed separately from the Note/Key and Gate CV.



Curve CV values (upper window).

This is useful for programming CV curves that control other parameters other than note pitch (although you could do this too). This way you could control the note pitch and triggering from the Key and Gate outputs for a device, then add a second independent pattern using the Curve CV output that could control filter cutoff for example.

It should be stressed that all three outputs can be used in any number of ways. For example, you could use the Gate CV to trigger a drum in Redrum, or let the Curve CV control the feedback parameter of a delay, etc.

Programming Patterns

! Pattern programming basics is covered in “Programming Pattern Devices”.

Tutorial

The programming procedure of the Matrix is to input note and gate values into the upper and lower fields of the pattern window respectively. You can input values by clicking or dragging in the pattern window. Proceed as follows:

1. Create a Subtractor synth.

You don't have to use the Subtractor device to use the Matrix, in fact you don't have to use an instrument device at all, but for this basic tutorial we will use a “standard” setup.

2. With the Subtractor selected, create a Matrix Pattern Sequencer.

The Matrix Note and Gate CV outputs will now be auto-routed to Subtractor's Sequencer Control Gate and CV inputs, as you can see if you flip the rack around.



3. Make sure that the switch to the left of the pattern window is set to “Keys” position.

As you can see, there is a horizontal row of red rectangles at the bottom of the upper field in the pattern window. These rectangles represent note pitch, for each step in a pattern. At the moment they are all set to the same note pitch.



4. Click inside the upper grid section of the Matrix pattern window.

An orientation line is displayed in the grid to make it easier for you to find the desired note, and the red rectangles are placed according to where you click. You can drag to input continuous note values.



5. Click in the lower area of the pattern window.

Vertical strips of varying heights can be created. These represent Gate velocity values. The higher the strip, the higher the velocity value. You can drag to input continuous gate values.



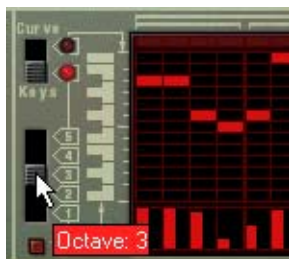
6. Press the Play button on the Matrix.

The pattern you “programmed” in the previous steps is now repeated. At the top of the pattern window, a red rectangle indicates every step of the pattern.

- If you now click or drag in the upper grid section with the pattern playing, you can hear how the note pitches change.

The note pitch corresponds to the keyboard printed to the left of the pattern window, in a one octave range, and as previously mentioned, an orientation line is visible when clicking or dragging, making it easy to find the note pitch on the keyboard.

- If you now click or drag in the lower gate section while the pattern is playing, you can hear how the timbre and volume changes.
- If you drag some of the vertical rectangles down so that they disappear from view, the corresponding steps of the pattern are completely silenced.
- By using the 5-way switch below the “Keys/Curve” switch you can input notes in other octave ranges (over five octaves).
Note that there can only be one note for each step in the pattern.



- By using a combination of the methods described in the above steps, you can program suitable note values for each step, decide which steps should be played and set their velocity with the gate values.

Using Curve Patterns

Curve patterns are independent patterns that can be applied separately to the note pattern programmed in “Keys” mode. If you switch the Keys/Curve switch to “Curve”, the note, but not the gate steps, disappear from view, and leaves the upper area of the pattern window empty. You can now start programming a curve pattern. Proceed as follows:

1. Draw a curve, using the same method as for notes or gates.

As you can see, the Curve pattern looks like large vertical gate steps.

- If you play the pattern, nothing has changed, i.e. the pattern sounds exactly like it did before the Curve pattern was drawn.

This is because the Curve CV output hasn't been connected to any parameter yet.

2. Flip the rack around so you can see the back panel of the Matrix.
3. Connect the Curve CV output to the Filter Cutoff Modulation Input on the Subtractor.

Now the curve pattern controls the filter frequency of the Subtractor.

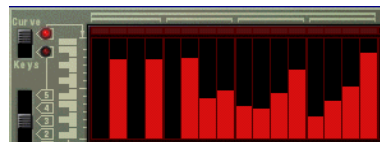
- ⚙ If the effect isn't very noticeable, try raising the filter Q parameter, and lowering the filter frequency.

- The Curve CV output can be connected to any device CV or Modulation input.

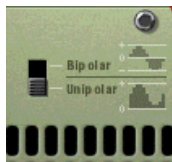
Actually, Curve CV signals can also produce Gate triggers (used for triggering samples or envelopes for example).

- A Gate trigger is produced for each curve pattern step *that follows a value of “0”*.

If you look at the picture below, steps 2, 4 and 6 will produce a trigger, because steps 1, 3 and 5 are set to zero, but the rest of the pattern would not.

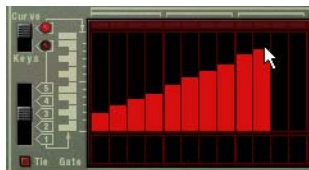


About Unipolar and Bipolar Curves



On the back panel of the Matrix you will find a switch, allowing you to select between “Unipolar” or “Bipolar” Curves. The difference is as follows:

- **A unipolar curve has values starting from “0” and up.**
“0” is the value produced by all steps when they are “empty” (not visible). Unipolar is the default setting of this switch when a new Matrix is created.



Unipolar curve.

- **A bipolar curve is divided in the “middle”, with the middle representing a value of “0”.**
The curve reflects this. If no curve has been drawn and you switch to bipolar mode, all steps go from the bottom up to the middle of the scale printed to the left of the pattern window. Thus, all steps are at “0”, and the curve can be drawn both up and down from the middle.



Bipolar curve.

Bipolar curves are essential in some instances. If you want to use the Matrix to CV control the Pan parameter for a mixer channel for example, a unipolar curve would start at zero - which for Pan equals center position. This means that you would only be able to use the curve to pan in *one direction* from this center position. A bipolar curve however, will have the zero value in the middle, allowing you to draw pan curves in both directions. Bipolar curves can also be used for controlling parameters with positive and negative values.

Setting Pattern Length



You may want to make settings for Pattern length, i.e. the number of steps the pattern should play before repeating:

- **The “Steps” spin controls are used to set the number of steps you wish the pattern to play.**
The range is 1 to 32. You can always extend the number of steps at a later stage, as this will merely add empty steps at the end of the original pattern. You could also make it shorter, but that would (obviously) mean that the steps you remove won't play back. The steps you remove aren't erased though, if you set the step number back again, anything recorded in the previously removed step locations will be played back.

Using Tied Notes

If you activate “Tie” to the left of the Gate pattern window, you can create longer notes (eighth notes, quarter notes etc.). A quick way to draw tied gates is to hold down [Shift] when you input the gate values.



Entering tied gate values.

- **Each step that has one tied gate value will be twice the length compared to a normal step.**
Tied gate steps are indicated by being twice as wide in the pattern window.
- **If two or more notes of the same pitch are tied together, the result will be even longer notes.**

Tied notes are also essential if you want to create typical TB-303 “Acid”-type lead lines - see [page 204](#).

Selecting Patterns and Pattern Banks

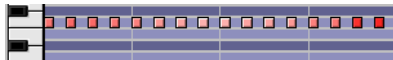
This is described in the chapter “Programming Pattern Devices” in the Getting Started book.

Setting Pattern Resolution

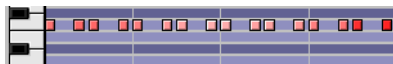
Matrix always follows the tempo setting on the transport panel, but you can also make Matrix play in different tempo “resolutions” in relation to the tempo setting. This is explained in the chapter “Programming Pattern Devices”.

Pattern Shuffle

Shuffle is a rhythmic feature, that gives the music a more or less pronounced swing feel. It works by delaying all sixteenth notes that fall in between the eighth notes.



Straight sixteenth note pattern (viewed in the sequencer).



The same sixteenth note pattern with shuffle applied.

In Reason, you can activate or deactivate shuffle individually for each pattern in a pattern device. However, the amount of shuffle is set globally with the Pattern Shuffle control on the transport panel.



The Shuffle on/off switch in Matrix and the Pattern Shuffle control on the transport panel.

Pattern Mute

If you deactivate the “Pattern” button above the Pattern select buttons, the pattern playback will be muted, starting at the next downbeat (exactly as if you had selected an empty (silent) pattern). For example, this can be used for bringing different pattern devices in and out of the mix during playback.

Pattern Functions

When a pattern device is selected, you will find some specific pattern functions on the Edit menu (and on the device context menu).

Shift Pattern Left/Right

The Shift Pattern functions move the notes and corresponding gate values in a pattern one step to the left or right.

Shift Pattern Up/Down

! This function does not alter the Curve CV. This is because the values produced by the Curve CV do not necessarily correspond to semitone note steps at all.

The Shift Pattern functions will transpose all the notes in a pattern one semitone up or down.

Randomize Pattern

The Randomize Pattern function create random patterns. These can often be great starting points and help you get new ideas. Both Note, Gate and Curve CV values will be created.

Alter Pattern

The Alter Pattern function modifies existing patterns. Note that there must be something in the pattern for the function to work on - using an Alter function on an empty pattern will not do anything.

! Note that Randomize and Alter affects both the Gate, Note and Curve CV!

Chaining Patterns

! Selecting Patterns and Banks and using Cut, Copy and Paste with Patterns is described in the chapter “Programming Pattern Devices”.

When you have created several patterns that belong together, you will most probably want to make these play back in a certain order.

→ **Simply activate record for the track with the Matrix as the destination in the sequencer and use the Pattern and Bank buttons to determine the playback order as the Song is playing.**

The Patterns play to the end before changing, so you won't have worry too much over the “timing” of the pattern changes you input manually. When you are done, the sequencer track will contain pattern change data, and the patterns will automatically switch according to the order set while recording.

4. More on recording pattern changes on [page 10](#).

→ An alternative way to do this is editing directly in the Pattern Edit lane in the sequencer.

Editing in the Pattern lane is described in the Sequencer chapter.

Converting Pattern Data to Notes

! Curve patterns cannot be converted to sequencer data! Only the note pattern and the gate values will be converted.

You can convert Matrix Pattern data to note data, that can be edited and played back from the main sequencer. Proceed as follows:

- 1. Select the sequencer track connected to the Matrix.**
- 2. Set the left and right locators to the desired range or length.**
If the range set is longer than the pattern(s), the data will be repeated to fit the range.
- 3. Select the Matrix device you wish to copy the pattern(s) from.**
- 4. Select “Copy Pattern to Track” from the Edit menu or the device context menu.**
Notes will be created between the left and right locators, according to the selected pattern (Gate and Key values only).

However, at this point the track with the notes is connected to the Matrix itself. This is pointless, since the Matrix doesn't produce any sound. Therefore:

- 5. Re-route the sequencer track to the device which was controlled by the Matrix (or to another instrument device if you like).**
This is done by clicking in the Out column for the track in the track list, and selecting another device from the pop-up menu that appears.

If you now activate playback from the transport you will send note data to the connected device from both the sequencer and the Matrix at the same time, which is probably not what you want. To avoid this happening, you have to do one of the following:

→ **Delete the Matrix device.**

Or...

→ **Disconnect CV and Gate cables between the Matrix and the instrument device on the back panel.**

★ **The procedure above copies a single pattern to notes in the sequencer. If you have automated pattern changes, you can copy a complete pattern track to notes, taking all pattern changes into account. This is described on [page 13](#).**

Example Usage

As mentioned previously, the Matrix is a very flexible device. Here follows a few examples of how you can use the Matrix Pattern Sequencer.

Using the Matrix for Modulation

You can effectively use the Matrix as a modulation source, much like an LFO. Just like the LFOs in Reason's instrument devices, the Matrix can generate modulation that is *synchronized to tempo*, which has many advantages. Proceed as follows:

1. **Create a Synthesizer (Subtractor or Malström).**
2. **Create a Matrix Pattern Sequencer, or if one already exists, set it to an empty pattern.**
These two devices may or may not be connected (by autorouting) via the synthesizers Sequencer Control inputs - it doesn't matter for this example.
3. **Flip the rack around and connect the Curve CV output on the Matrix back panel to the "Amp Level" Modulation input on the synthesizer.**
This parameter is used for modulating the output level (volume) of the synthesizer. Volume modulation is often referred to as Tremolo. You can use a unipolar curve (see [page 201](#)) for this example.
4. **Flip the rack back again, and switch the Matrix to display the Curve pattern window.**
You should now see an empty pattern window, with no Gate or Curve events visible.
5. **Draw a curve like the one shown in the illustration below.**
If you use fewer or more steps than 16 (as shown in the picture), just draw the curve so that it roughly matches the shape in the picture.



6. **Activate Click on the transport panel.**
7. **Select the track that is routed to the synthesizer, so that you can play it from your MIDI keyboard.**
8. **Activate Play on the transport panel, and hold a chord down on your keyboard.**
You should now hear the volume being modulated by the Curve pattern.

9. **While still in play mode, you can use the Resolution knob to change the modulation "rate" in relation to the tempo.**

For each clockwise resolution step the modulation speed is doubled and vice versa, but it will always stay in sync with the tempo.

Programming "Acid Style" lead lines

By "acid style" lead lines we mean patterns that use a combination of Legato and slide (or portamento) effects to produce the widely used hypnotic "wavy" sound produced by the original Roland TB-303, and recreated in the Propellerhead Software product ReBirth. To approximate this typical sound using Reason, proceed as follows:

1. **Create a Synthesizer (Subtractor or Malström).**
 2. **Create a Matrix Pattern Sequencer, or if one already exists, set it to an empty pattern.**
 3. **Make sure that the Note and Gate CV outputs are connected to the synthesizers Sequencer Control CV and Gate inputs, respectively.**
 4. **For Subtractor, select either a Init Patch, or use the "TB Synth" patch in the Monosynth category of the Factory Sound Bank.**
 - **If you use an Init patch, it is important that you make the following settings:**
 - Set Polyphony to "1".
 - Switch Trigger Mode to "Legato".
 - Set Portamento to a value around "50".
 5. **Create a pattern in Matrix, and keep it playing back.**
 - **If "Tie" (see [page 201](#)) now is activated for a step, the note will be tied to the next and the pitch will continuously "glide" to the pitch of the following step.**
Please note that Tie should be activated on the note you wish to slide from, and not the note you slide to.
 - **If you have several tied notes, one after the other, they will play as one long legato phrase. This can be used to create "wavy" lead lines with pitch bend effects.**
 6. **Experiment with different Note, Tie and Gate values.**
If you have ever used a TB-303 or ReBirth, you should now begin to get the hang of how you can create patterns in that particular style by using the Matrix together with a synthesizer.
- ★ **Adding a DDL-1 (delay), and a D-11 (distortion) effect device will make it sound even more "ReBirth"-like, but of course you are also able to get a much wider range of timbres by utilizing Reason's other sound and modulation capabilities.**

Triggering Samples

The Gate CV output can be used to trigger samples, either in Redrum or in the NN-19 or NN-XT Sampler.

- **Connect the Matrix Gate CV out to the Gate (Sequencer Control) in on the NN-19/NN-XT or to one of the individual Gate Channel inputs of Redrum.**

The Matrix gate values will now trigger the sample on each step that has a Gate value above "0".



REASON

18

→ ReBirth Input Machine

Introduction



The ReBirth Input Machine is a device dedicated to receiving audio from the Propellerhead program “ReBirth RB-338” (version 2.01 and later). This is achieved by using ReWire technology (see [page 48](#)), where Reason will act as master and ReBirth as a slave device. If you don’t have ReBirth installed, you cannot use this device. If you are a ReBirth user, you can use the ReBirth Input Machine for the following:

- ➔ **Receive up to eighteen channels of streaming ReBirth channels in Reason.**
You can create more ReBirth Input Machines, but only one can be active at a time.
- ➔ **Sample accurate synchronization between the audio in the two programs.**
- ➔ **The two programs can share the same audio card and take advantage of multiple outputs on that card.**

Preparations

For the ReBirth Input Machine to correctly operate together with ReBirth, the launch and quit order is very important. Proceed as follows:

Launching

1. **Launch Reason.**
2. **Create a ReBirth Input Machine.**
You may want create a Mixer prior to this step, otherwise the L/R Mix channels will be routed directly to the Audio Hardware Interface. If you have a Mixer, the L/R Mix output from the ReBirth Input Machine will be automatically connected to the mixer’s first available audio inputs.
3. **Launch ReBirth.**
4. **When ReBirth is launched, select Reason as the application in focus.**
If both the “Reason is Rewire Master” and the “Active” indicator on the ReBirth Input Machine are lit, this indicates that the launch procedure was correct and that Reason and ReBirth are now locked and in sync.

- ➔ If only the “Active” indicator is lit, either the launch order was wrong, or ReBirth is not installed properly.
- 5. **Activate playback on Reason’s transport panel.**
ReBirth and Reason are locked in perfect sync, and will follow any transport commands in either of the programs.
- ! **Note that there is no master/slave relationship for the transport controls when using ReWire, as either device will control the other device’s transport. The audio, however, is streamed from ReBirth to Reason, so in this aspect Reason is the master device.**

Quitting

1. **First quit ReBirth.**
2. **Then quit Reason.**

Routing

When the two programs are synced, you can route any of the eighteen available outputs in ReBirth, to separate channels in a Reason Mixer, or to the Hardware Interface for direct connection to a physical output on your audio card.

If you flip the rack around, a row of 18 audio outputs is shown, with the L/R Mix outputs auto-routed to your mixer or to the hardware interface.

What Signals are on the Outputs?

Mix-L and Mix-R

This is the regular master output in ReBirth RB-338. These are the only stereo channels, all other channels are in mono.

- ➔ **If none of the other channels are used, then this carries all the sound from ReBirth.**
- ➔ **Signals that are activated separately are removed from this mix.**
If for example the 909-Mix channel is activated, then Mix-L and Mix-R carries all the sound from ReBirth RB-338 except the 909, which will appear on its own channel.

The individual outputs are described more closely in the ReWire chapter of the ReBirth manual.



REASON

19

→ BV512 Vocoder

Introduction



The BV512 is an advanced vocoder device with a variable number of filter bands. It also has a unique 1024-point FFT vocoding mode (equivalent of 512-band vocoding) for very precise and high quality vocoded speech. By connecting the BV512 to two instrument devices, you can produce anything from vocoded speech, singing or drums to weird special effects.

Even if you have worked with a vocoder before, please read the following section. Knowing the basic terms and processes will make it much easier to get started with the BV512!

How does a vocoder work?

Carrier and modulator

A vocoder accepts two different input signals, a “carrier” and a “modulator”. It analyzes the modulator signal, applies its frequency characteristics to the carrier signal and outputs the resulting “modulated” carrier signal.

In the most typical case, the carrier signal is a string or pad sound and the modulator signal is speech or vocals - the result will be a talking or singing synth sound. The modulator could also be drums or percussion (for rhythmically modulated sounds and effects) or any sound with changing frequency content.

Filter bands

Technically, a vocoder works in the following way: The modulator signal is divided into a number of frequency bands by means of bandpass filters (called the “modulator filters” or “analyzing filters”). The signal in each of these bands is sent to a separate envelope follower (which continuously analyzes the level of the signal). The carrier signal is sent through the same number of bandpass filters (the “carrier filters”), with the same frequency ranges as the filters for the modulator signal. The gain of each bandpass filter is controlled by the level from the corresponding envelope follower, and the filtered signals are combined and sent to the vocoder’s output.

In this way, the carrier is filtered to have roughly the same frequency characteristics as the modulator. If the modulator signal has a lot of energy in one of the frequency bands, the gain of the corresponding filter band for the carrier signal will be high as well, emphasizing those frequencies in the output signal. If there is no signal at all within a frequency band in the modulator signal, the corresponding band in the output signal will be silent (as the gain will be zero for that filter).

There are several factors determining the quality of the vocoder sound, but the most important is the number of filter bands. The larger the number of filter bands, the closer will the output signal follow the modulator’s frequency characteristics. The BV512 offers 4, 8, 16 or 32-band vocoding.

❗ **Even if a high number of bands will make the sound more precise and intelligible, this isn’t always what’s desired! Vocoding with a lower number of bands can give results that sound different, fit better in a musical context, etc.**

FFT vocoding

The BV512 has an additional FFT mode, in which the vocoding process isn’t based on bandpass filters as described above. Instead, FFT (Fast Fourier Transform) analysis and processing is used. This equals 512 “conventional” frequency bands and results in a very precise and detailed vocoder sound. Note:

- The FFT mode is best suited for vocoding speech or vocals, giving crystal clear and highly intelligible results. It is not so well suited for vocoding drums and percussion, since the FFT process is inherently “slower” than the regular filtering and doesn’t respond as quickly to transients, and also there will be a slight delay added to the signal (in the region of 20ms). A workaround solution to this would be to move the modulator signal slightly ahead to compensate for the delay.
- Where the conventional filter bands are distributed logarithmically (i.e. the same number of filter bands per octave), the 512 bands in the FFT mode are distributed linearly. This means a lot of the bands will be in the high frequency range - this is one of the reasons for the clear sound but it is also something to keep in mind when making settings for the vocoder in FFT mode.

Setting up for basic vocoding

This tutorial describes how to connect and use a typical vocoder setup. We assume here that you have a MIDI keyboard connected. For details on the parameters, see [page 214](#).

1. **Make sure there's a Mixer device in the rack (with at least one free channel).**

2. **Create the instrument device you want to use for the carrier signal.**
This could typically be a synth or a sampler. In this example we choose a Subtractor synthesizer.

3. **Set up the carrier device for a sustaining, bright sound.**
It's important to have high frequencies in the carrier. On the Subtractor, a simple but effective carrier sound would be based on a sawtooth wave, with the filter fairly open. For more about choosing carrier sounds, see [page 216](#).

4. **Select the carrier device and create a BV512 Vocoder.**
If you flip the rack around you will see that the Vocoder is automatically routed as an insert effect for the carrier device (using the Carrier Input jacks).



5. **Press [Shift] and create the instrument device you want to use for the modulator signal.**

Pressing [Shift] will add the device without auto-routing it to a mixer - this makes sense since we want to route it to the Vocoder in this case. For a modulator device you would typically either want a sampler (with vocals or speech samples), a drum machine or a Dr.Rex device (with vocal or rhythmic loops). For simplicity we use a Dr.Rex device in this example.



6. **Flip the rack around and route the output of the Dr.Rex to the Modulator Input jack on the BV512.**
7. **On the BV512 Vocoder, turn the Dry/Wet knob fully to the left ("dry").**



This will let you hear the unprocessed sound of the modulator device only - useful for the next step:

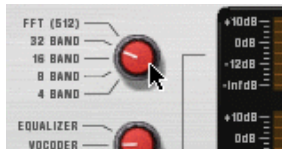
8. **Load a loop into the Dr.Rex device and click the Preview button to start playback.**
For example, you could simply choose one of the Dr.Rex Drum Loops in the Factory Sound Bank.
9. **Turn the Dry/Wet knob on the vocoder fully to the right ("wet").**
Now you won't hear anything - since there is no carrier signal.

10. Route MIDI to the carrier device by clicking in the MIDI symbol column for its track in the sequencer.

11. Play a chord or a note on your MIDI keyboard.

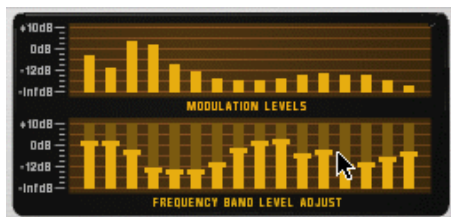
What you hear now is the vocoded sound, e.g. the carrier sound processed to have the same tonal characteristics as the modulator.

12. Try the different filter band options and note the difference in sound.



13. You can also adjust the vocoder sound by clicking and dragging the bars in the lower display.

Each bar corresponds to a frequency band, with low frequencies to the left and high frequencies to the right. You adjust the level of a band by dragging its bar up or down. Clicking and dragging across the bars allow you to change the levels of several bands, much like drawing an eq curve.



The upper display shows the spectrum of the modulator signal, for display only.

→ **To reset a band to ± 0 dB, press [Command] (Mac) or [Ctrl] (Win) and click on it.**

You can also reset all bands to zero by bringing up the context menu for the Vocoder device and selecting "Reset Band Levels".

14. If the vocoder sound is "muddy" or indistinct, try raising the "HF Emph" knob on the Vocoder.

This parameter (High Frequency Emphasis) boosts the high frequencies in the carrier signal.

15. Try out the other parameters if you like.

See [page 214](#) for details.

That's it - a basic vocoder setup!

Vocoded vocals

The most common usage for a vocoder is probably the typical "singing" or "talking synth" sound, using vocals or speech as modulator. Since Reason doesn't support live audio input you cannot sing and play in real time - instead you need to use sampled speech or vocals (with e.g. an NN-19 or NN-XT as the modulator device). The procedure for this is roughly the same as in the tutorial above, but this time you need to record or enter some notes in the sequencer for the modulator device (since the samplers don't have pattern or Preview playback). Here's a quick guideline:

1. Create the carrier device.

2. Select the carrier device and create a BV512 vocoder.

3. Select the BV512 and create the modulator device (typically an NN-19 or NN-XT sampler device).

4. Load the vocals or speech samples into the sampler device and assign them to keyzones as desired.

For details about using sampler devices, see the respective device chapter.

5. Record or enter some notes on the sequencer track for the sampler device, so that the vocal samples are played back where you want them in the song.

To hear the unprocessed sound of the sampler device, set the Dry/Wet control on the BV512 to "Dry", as above. When you're done, turn the control back to "Wet" to get the vocoded sound.

6. Route MIDI to the carrier device.

7. Start sequencer playback and play notes or chords on your MIDI keyboard.

The result will be the classic vocoded vocal sound.

8. At this point you may want to record the notes or chords you play for the carrier device.

As MIDI is already routed to the carrier device track, all you need to do is start recording and play along.

Using the BV512 as an equalizer

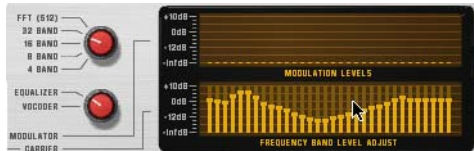
The BV512 has a unique equalizer mode, in which the device works purely as an insert effect (the modulator input isn't used). This allows you to use the processing filters of the vocoder as a kind of graphic equalizer.

Setting up

1. **Select the device that you want to process through the BV512.**
2. **Create a BV512 device.**
It is automatically connected as an insert effect, using the Carrier Input jacks.
3. **Set the switch to the left of the displays to "Equalizer".**



In use



In equalizer mode, you cut or boost frequencies by clicking and dragging in the lower display - just as with a regular graphic equalizer. The usage and results differ depending on which mode is selected:

4 - 32 band mode

As in vocoder mode, the number of bars in the display conforms to the number of bands selected (4, 8, 16 or 32). With a higher number of bands you get a more detailed control over the frequency response. However:

- ➔ **In these modes, the equalizer will "color" the sound even if all bands are set to ± 0 dB!**
This is due to phase interaction and overlap between the bandpass filters.

Therefore you probably want to use the 4 - 32 band mode for coloring and mutating sounds - not for subtle, "clean" equalizing.

FFT (512) mode

In FFT (512) mode you still get 32 bars in the display, but each bar may control several frequency bands (remember that there are 512 bands in FFT mode). Since the frequency bands are distributed linearly in FFT mode, bars to the left in the display control few frequency bands while bars to the right control many frequency bands.

- ➔ **In FFT (512) mode, setting all bands to ± 0 dB is the same as bypassing the equalizer - the sound will not be affected.**
This makes FFT mode suitable for "clean" equalizing, where you want to boost or cut some frequencies without changing the basic sound character.
- ➔ **However, FFT mode equalizing is not suited for very drastic frequency cuts or boosts, as this may give audio artefacts due to the workings of FFT processing.**
Still: as always, there are no hard and fast rules. Let your ears judge!
- ➔ **Keep in mind that FFT mode also introduces a slight delay to the signal.**

BV512 parameters

On the front panel of the BV512 Vocoder, you will find the following parameters and displays:

Parameter	Description
Bypass/On/Off switch	In Bypass mode, the carrier signal passes through the device unaffected and the modulator signal is disregarded. In On mode, the device outputs the vocoded or equalized signal. Off mode cuts the output, silencing the device.
Level meters	Show the signal level of the carrier and modulator signals, respectively.
Band switch	Selects the number of filter bands (4, 8, 16 or 32) or FFT (512) mode.
Equalizer/Vocoder switch	Determines whether the BV512 should work as a vocoder or an equalizer. In Equalizer mode, the Modulator input is disregarded (see page 213).
Modulation level display	The upper display shows the spectrum of the modulator signal.
Frequency band level adjust	The lower display allows you to adjust the level of each filter frequency band, by clicking and dragging the corresponding bar. In vocoder mode this affects the vocoded sound. In equalizer mode, this is where you cut or boost frequencies. To reset a band to ± 0 dB, press [Command] (Mac) or [Ctrl] (Win) and click on its bar in the display. To reset all bands, select "Reset Band Levels" from the device context menu. Note: when FFT (512) mode is selected, each of the 32 bars in the display corresponds to several frequency bands, with bars to the right in the display controlling progressively more bands (due to the FFT bands being linearly distributed over the frequency range).
Hold button	Clicking this button "freezes" the current filter settings. While the button is lit, the modulator signal doesn't affect the sound - the carrier signal is filtered with the settings as they were the moment you activated Hold. Click the button again to turn off Hold. Hold is also automatically reset (turned off) when you stop sequencer playback - just like the pitch bend and modulation wheels on synth devices. This function can be controlled via CV or MIDI, for sample and hold-like effects. The Hold button is not available in Equalizer mode.

Parameter	Description
Attack	This is a global attack time control, affecting all envelope followers (see page 210). Normally you probably want this set to zero, to make the vocoder react as quick as possible. Raising the Attack time can be useful for "smearing" sounds, creating pads, etc. Not available in Equalizer mode.
Decay	Similarly, this controls the decay time for all envelope followers, i.e. how quick the filter band levels drop. Adjust this according to taste and context. Not available in Equalizer mode.
Shift	Shifts the carrier filters up or down in frequency, drastically changing the character of the vocoded (or equalized) sound. This parameter can be controlled via CV, for phaser-like sweeps and special effects.
HF Emph (High Frequency Emphasis)	Boosts the high frequencies in the carrier signal. This is sometimes desired to get a clearer vocoded sound. The reason is that a carrier signal should theoretically contain roughly equal energies in all frequency ranges for best results - in a typical synth sound the high frequencies are often weaker than the low frequencies. Raising the HF Emph control will rectify this. Not available in Equalizer mode.
Dry/Wet	Determines the balance between modulator sound (dry) and vocoded sound (wet). To get the pure vocoder sound, set this to wet (turned fully right). Not available in Equalizer mode.

Connections



The back panel of the BV512 offers the following connections:

Individual band levels

These are CV outputs and inputs.

- The upper row outputs CV signals generated by the envelope followers for each frequency band.
- The lower row are CV level inputs to the individual bandpass filters through which the signal is processed (the “vocoder filters”). Connecting a CV signal to one of the inputs breaks the internal signal path from the corresponding envelope follower (in other words, that frequency band is now controlled by the CV signal you’ve connected - not by the corresponding frequency band in the modulator signal).
- If 16 band mode is selected, each output/input pair corresponds to a separate frequency band. In 8 band or 4 band mode, only the 8 first or 4 first output/input pairs are used. In 32 band mode, each output is a mix of two adjacent frequency bands and each input controls two bands. Finally, in FFT (512) mode each output/input pair corresponds to several frequency bands.

There are several interesting uses for the Individual band levels connectors: you can cross-patch frequency bands so that e.g. low frequencies in the modulator signal controls high frequency bands in the vocoder, you can extract CV signals for controlling synth parameters in other devices, you can base the vocoding on CV signals from other devices rather than on a modulator signal, etc. See [page 218](#) for details.

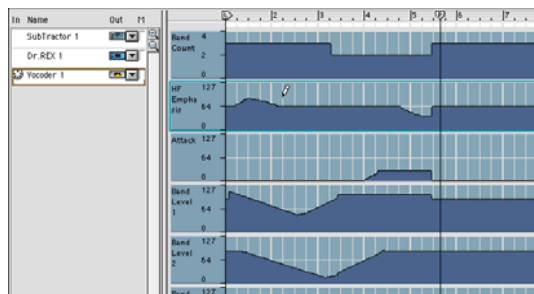
Other CV connections

Connection	Description
Shift (CV in)	This allows you to control the Shift parameter from an external CV source. A sensitivity knob determines how much the Shift setting is affected by the CV signal.
Hold (Gate in)	When a gate signal is sent to this input, the Hold function is activated (see page 214). Hold remains on until the gate signal “goes low” (falls to zero). By connecting e.g. a Matrix to this input, you can create “stepped” vocoder sounds, sample and hold-like effects, etc.

Audio connections

Connection	Description
Carrier input	This is where you connect the instrument device that provides the carrier signal (or the device to be processed in Equalizer mode) - typically a synth or sampler device. The vocoder can handle mono or stereo carrier signals.
Modulator input	This is where you connect the instrument device that provides the modulator signal, in mono. This connection is not used in Equalizer mode.
Output	In Vocoder mode, the outputs carry a mix between the vocoded signal and the modulator signal (as set with the Dry/Wet control on the front panel). In Equalizer mode the output is the carrier signal, processed through the equalizer filter. Note that the output will be in mono if the Carrier input is in mono, and vice versa - the BV512 does not process mono into stereo.

Automation

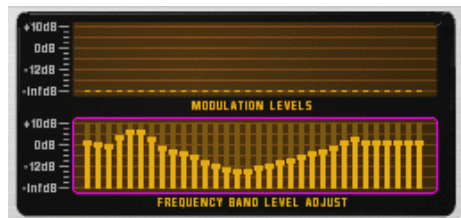


All parameters on the front panel can be automated in the standard manner. The individual band levels (the bars in the lower display) will be edited on separate lanes in the sequencer. Note:

→ **As with the other effect devices, you have to manually create a sequencer track for the BV512.**

→ **Although the band level adjustments can be edited individually, they are treated as one automatable parameter on the device panel.**

This means that if any single band level control is automated, there will be a frame around the whole lower display on the device panel. [Ctrl]-clicking (Mac) or right-clicking (Win) in the lower display and selecting "Clear Automation" will remove the automation for all bands. Similarly, selecting "Edit Automation" will open the sequencer with lanes for all band levels shown.



The frame indicates that one or more band level controls are automated.

Tips and tricks

Choosing a carrier sound

As always, which carrier sound to choose is a matter of taste and musical context. However, here are a few guidelines to help you get a good result:

- The carrier sound should preferably have a lot of harmonic content (brightness) - dark or muffled sounds will not "give the vocoder much to work with".
- Often, you want the carrier sound to sustain at an even level (i.e. it shouldn't "die out" when you hold a chord). Similarly, you most often want a reasonably fast attack (although not with a distinct, sharp click or edge).
- You may want a sound that is rather static over time, without drastic envelope control of filter cutoff for example.
- If you want to play vocoded chords, the carrier sound must of course be polyphonic.

Here are some hands-on suggestions for carrier sounds:

→ **A simple Subtractor pad based on a sawtooth waveform.**

You could simply start with the initial patch (as set up when you create a new Subtractor device). Open the filter, turn off envelope modulation of the cutoff frequency and raise the Amp Envelope Sustain.

If you want a classic, rich chorus-like sound, use two detuned oscillators - or better still, add a UN-16 Unison device as an insert effect between the Subtractor and the vocoder!



A simple but effective carrier sound setup.

→ **A similar fat carrier sound can be obtained using a Malström device with a patch based on the "Sawtooth*16" graintable.**

With the Malström you can get a stereo carrier signal with no extra devices: simply select the "Sawtooth*16" graintable for both oscillators, detune the oscillators slightly with the Cent controls and raise the Spread parameter to the desired stereo width. No filter routings are necessary.

- **For a more distinct and precise sound, try using a narrow pulse waveform.**

You get this by selecting e.g. a sawtooth wave on the Subtractor, setting the Phase Mode selector to “-” and turning the Phase knob to the left until you get the desired sound. This type of carrier sound lends itself well to monophonic vocoder lines in the lower registers.

- **Use noise as a carrier.**

Try using pure noise (possibly filtered down a bit) for robotic voices, whispering and special effects. It’s also very useful to add a bit of noise to a sawtooth or pulse sound - this makes vocoded speech clearer and more intelligible.

- **Use sampled strings or choir sounds.**

A rich drawbar organ sample can also be a cool carrier sound.

- **For unusual vocoder sounds, try using the Malström as carrier device, e.g. with a glassy, digital pad sound selected.**

Try turning up the Attack and Decay controls on the BV512, for smeared, rhythmic or pseudo-random modulation of a pad.

Choosing a modulator sound

The modulator sound should typically have varying level and harmonic content. As we’ve already mentioned, the most typical modulator sounds are vocals or speech and drums or percussion.

- **The quickest way to get a modulator sound is to use a rhythmic loop in the Dr.Rex device (as in the tutorial on [page 211](#)).**

This way you don’t have to program a rhythm pattern. On the other hand, using a Redrum as modulator allows you to create exactly the rhythm you want and fine-tune the sounds and the groove.

- **To use your “own” vocals as modulator sounds, you need to record them as WAV or AIFF files (using any audio recording utility on your computer) and load the files as samples into an NN-19 or NN-XT device.**

- **Instead of using a sampler device as modulator for speech or vocals, you could slice the vocal samples in Propellerheads’ ReCycle application, and play them back with a Dr.Rex device.**

This would make it easier to work with vocoded vocals, especially if you are experimenting with different tempo settings or grooves. Tip: You can copy the MIDI notes played by Dr.Rex to the carrier track so that the original rhythm of the speech/vocal is preserved.

Using the modulator as carrier

You can get cool special effects by using the same device both as carrier and as modulator. For example, try processing a Redrum device in the following way:

1. **Create a Redrum device and set up the desired patch and pattern.**
2. **Create a Spider Audio Merger & Splitter device.**
3. **Create a BV512 Vocoder.**
4. **Flip the rack around and connect the devices in the following way:**



The output of the Redrum goes into the splitter section of the Spider, and is split into two signals. One signal goes into the carrier input of the vocoder, the other goes into the modulator input.

This is essentially the required connections, but for best results it’s a good idea to add some distortion and/or compression to the carrier signal - this increases the amount of high frequencies in the carrier signal:

5. **Press [Shift] and create a Scream 4 distortion device.**
 6. **Connect the distortion device as an insert effect between the Spider and the carrier input of the vocoder.**
Now, the carrier signal will be processed in the distortion device, but not the modulator signal.
 7. **Play back the pattern and experiment with the settings on the vocoder and distortion device.**
- **This technique can also be used to process vocals and speech.**
- **Try adjusting the Shift parameter for new effects and sounds.**
Remember that you can route CV to the Shift parameter on the back of the BV512 - use e.g. a Matrix or an LFO output on a synth device!

Controlling the Hold function

As described see [page 214](#), pressing the Hold button on the front panel “freezes” the current filter spectrum until you deactivate it again. This can be used for creating sample & hold-like effects, stuttering or garbled vocoder sounds:

- Connect e.g. the Gate output on a Matrix device to the Hold input on the back of the BV512. By playing back a gate pattern on the Matrix, the Hold function will repeatedly be turned on and off according to the programmed rhythm in the pattern. Hold will be active for the length of each gate signal.
- Automate the Hold function with the main sequencer, either by recording it or by drawing in its controller lane.
- If you route MIDI to the BV512 you can control the Hold function in two ways by default: By pressing a damper pedal connected to your MIDI controller or by playing the note C4. In both cases, the Hold function will be momentary - Hold is on until you release the pedal or key.

Using the individual band level connections

As described on [page 215](#), the individual band level connectors on the back are CV output and input jacks. The upper row sends out the CV signals from the envelope followers for the different frequency bands, while the lower jacks are CV inputs for controlling the individual bandpass filters (breaking the internal connection from the envelope followers). There are several interesting things you can do with these connections:

Crosspatching frequency bands



By connecting outputs to inputs in alternative configurations, you can drastically change the result of the vocoding. For example, you could have low frequencies in the modulator signal give high frequencies in the vocoded sound and vice versa. Note:

- In 4 band and 8 band mode, only the 4/8 first output/input pairs are used.

- In 32 band mode and FFT (512) mode, each connection corresponds to two or several frequency bands.

This means that connecting an output to the input with the same number is *not* the same as using the internal signal path (no CV cable connected). You can hear this quite clearly in FFT (512) mode: connect all outputs to the corresponding inputs and gradually remove the CV cables while listening to the vocoder sound - the sound will progressively get more detailed.

Extracting CV from the vocoder

You can connect an individual band level output to any CV input on any device. This means you can use the vocoder as an envelope follower, having elements in the modulator sound control a parameter in another device, e.g. an effect. Note:

- The Attack and Decay settings on the BV512 panel affect the envelope followers, and thus the rise and fall times of the CV signals from the individual band level outputs.
- If you are using the vocoder in a mode with many bands, but want a broader frequency range to generate the CV signal, you can merge several band outputs into one CV signal - use a Spider CV Merger & Splitter device.

Controlling vocoder bands from an external source

Connecting a CV source to an individual band input breaks the internal connection from the corresponding envelope follower. This way you can “manually” control the vocoder filters. Some applications:

- **Connect the CV outputs for one or more envelopes in the carrier device to individual band inputs.**

When you play the carrier instrument, one or more of the bandpass filters in the vocoder will automatically open, adding an extra attack to the sound. Useful if you really want to “play” the carrier, rather than just hold a chord.

→ **Connect the gate outputs on a Redrum device to individual band level inputs.**

With this connection (and no device connected to the Modulator input), the Redrum will serve as a pattern sequencer, opening and closing different filter bands. To adjust the gate times, set the drum sounds to Gate mode and use the Length parameter. The result is totally different from using the audio signal of the Redrum as modulator.



The vocoder bands are now solely controlled by the gate signals from the drum channels - the modulator input isn't used.

Note that you can use a Spider CV Merger & Splitter device to split a gate signal, sending it to several bands. Also, note that the velocity of the programmed drum notes govern the level of the corresponding filter bands.

“Playing” the vocoder from a MIDI keyboard

If you have routed MIDI to the BV512, playing notes from C1 and up will control individual filter bands. For example, in 16 band mode, C1 controls band 1, C#1 band 2 and so on up to D#2 (which controls band 16).

- The level of the bands is proportional to key velocity (how hard you play).
- A band will be “open” until you release the corresponding key.
- Bands to which you have connected a CV signal (using the individual band level inputs on the back panel) will not respond to MIDI keys.

Note that with this function, you “play the modulator”. You still need a carrier signal to get any sound. Typically, you would first record the notes or chords for the carrier device in the sequencer, then route MIDI to the vocoder and “play” it from your MIDI keyboard while playing back the recorded carrier notes.

- ★ An interesting application of this is to patch the vocoder as an insert effect for the whole mix (the output of the main mixer connected to the carrier input, with no modulator device connected), and “play the vocoder”. Only the frequency bands for which you press keys will be let through. Use the FFT (512) mode for best results.

Using the BV512 as a reverb

This is a very special trick which can be quite cool. Proceed as follows:

1. Create a Redrum device.

The “vocoder-reverb” is best suited for drums, even though nothing stops you from using it on other sounds.

2. Create a Subtractor and a vocoder.

The Subtractor will automatically be routed to the carrier input. We don't need a dedicated modulator device in this setup.

3. Flip the rack around and connect Aux send 1 on the Mixer to the modulator input on the vocoder.

4. While you're there, re-route the vocoder output to Aux return 1. This way, our vocoder-reverb will be connected as a regular send effect.



5. Set the vocoder to FFT (512) mode, turn the Decay knob between 6 and 7 and turn the Dry/Wet control to “wet” (fully right)

6. On the Subtractor, set up a noise sound as follows:

Turn the Oscillator Mix knob fully to the right.
Turn on the Noise section (but make sure Osc 2 is off).
In the Noise section, turn Color to around twelve o'clock.
Open the filter fully and make sure resonance is set to 0.
Make sure Filter Envelope Amt is 0 (and turn off velocity modulation).
Raise the Sustain to full in the Amp Envelope section.



Now we want the Subtractor to play a continuous noise. You could just route MIDI to it, play a note and keep it pressed, but that will probably wear you out in the long run. Better to use a Matrix:

7. Create a Matrix and route it to the Subtractor.

We really only need the Gate connection - the note number isn't important with the noise patch.

8. Set up a one step pattern with a tied gate (press [Shift] and draw the gate) and start playback on the Matrix.

Now the vocoder gets a continuous noise signal as carrier.

9. Create a suitable drum pattern on the Redrum and start pattern playback.

10. Gradually turn up send 1 for the Redrum channel in the mixer.

This now serves as a balance control between the dry drum sound and the reverb, generated by the vocoded noise! Set it to a suitable reverb level.

11. Use the Decay control on the vocoder to adjust the reverb decay time.

12. Use the Noise Color control on the Subtractor to make the reverb darker or brighter.

You could use the filter cutoff for this as well.

That's it - a pretty good reverb sound with a lot of control. Although the settings above give the most natural sound, you can vary the sound and create special-FX reverb in the following ways for example:

- Switch the vocoder to a lower band mode.
- Lower the cutoff and add some resonance in the Subtractor filter.
- Modulate the Subtractor filter with a fast LFO.
- Set the Subtractor filter to HighPass mode to remove the bottom end from the reverb.
- Turn off the Matrix controlling the Subtractor and “play” the noise patch yourself (or from the sequencer). This way you can create gated reverb effects, etc.

Creating a stereo reverb

What you've got above is a mono reverb. Here's how to make it stereo:

1. **Select the Subtractor and create a Spider Audio Merger & Splitter device.**
2. **Create a DDL-1 delay.**
3. **Connect the devices in the following way:**

The Subtractor output should be routed to a Splitter input on the Spider. One split output should be routed to one of the carrier inputs on the vocoder, the other split output should be routed to the delay. The delay output (mono) should be routed to the other carrier input on the vocoder.



The vocoder will now get a "fake-stereo" noise carrier signal.

4. **Make sure the output from the vocoder is connected in stereo to the Aux return on the Mixer.**
5. **Finally turn down the Feedback on the delay, set it to all "wet" and set the decay time to a second or so.**

When you now start playback on the Redrum, the reverb will be in stereo!



REASON

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→ The Effect Devices

Common Device Features

While the specific parameters for each device are described below, some features and procedures are common to all effect devices:

The Input meter



This shows the level of the incoming audio signal, giving you an indication of which devices are active, connected and playing. However, you don't need to worry about clipping in effect devices, even if the meter goes into the red.

The Power/Bypass switch

This is located in the upper left corner of each effect device. The switch has three modes, according to the following figure:



Mode	Description
Bypass	In this mode, the input signal is passed directly to the audio output, without being affected by the effect device. This is useful when the effect device is connected as an insert effect, and you want to compare the effect sound with the dry sound.
On	This is the default mode, in which the device processes the incoming signal.
Off	In this mode, the effect device is turned off and neither dry nor effect sound is sent out. This is useful when the device is connected as a send effect and you want to turn it off temporarily.

About making settings

You adjust effect parameters using the regular editing techniques, as described in the Getting Started book. Note:

- A quick way to reset the parameters to their default values is to [Command]/[Ctrl]-Click the corresponding knob.

About Connections

- All effect devices have stereo inputs and outputs, and can be connected as send effects or as insert effects.

However, some effects are best used as one of these only. This is stated for each effect on the following pages. See also the section about the signal flow graphs below.

- Most of the effect devices have one or several CV inputs on the back panel.

These allow you to control various effect parameters in real-time, from another device in the rack. See [page 36](#) for details about routing CV.

The Signal Flow graphs



On the back of each effect device, you will find two or three small "graphs". These indicate how the effect device handles mono and stereo signals, depending on the connections. The selection of graphs for a device tells you how it should be used, according to the following rules:

Graph	Description
	Can be connected as a mono-in, mono-out device. (Of course, all effect devices <i>can</i> be connected in mono. However, if this graph isn't shown for a device, this means that a mono-in, mono-out connection may not give the proper results).
	Can be connected as a mono-in, stereo-out device. This means that the device creates some sort of stereo effect (e.g. a reverb) or a mono effect that can be panned.
	If you connect both inputs and outputs in stereo, the two sides will be processed independently (dual mono processing).
	If you connect both inputs and outputs in stereo, the two sides are summed before the effect processing. However, the actual effect is in stereo (and the dry signal will remain in stereo, if it is passed through the effect).
	"True stereo" processing, or "stereo in - stereo out" processing. When you connect the inputs in stereo, each channel in the effect uses the signal information from both inputs. However, the inputs are not summed - the two channels are processed differently. This mode is available on the RV7000 Advanced Reverb.

Scream 4 Sound Destruction Unit



Scream 4 is a very versatile stereo in/out sound destruction device, capable of warping any sound literally beyond recognition, but also capable of producing more subtle musical effects. Scream 4 features a wide range of algorithms for distortion and sound mangling which can be combined with an EQ and a resonant "Body" section to provide everything you need to add an edge to your sounds. This effect is most often used as an insert effect.

About the Patch format

Unlike most of the other effect devices, Scream 4 features programmable effect presets. Included are a number of factory Patches which can be used as they are or provide you with a good starting point for further tweaking.

Patches use the Windows file extension "*.SM4". Loading and saving Patches is done in the same way as for instrument devices.

Parameters

Scream 4 contains three main sections; Damage (distortion and other types of sound destruction), Cut (EQ) and Body (places the sound in a resonant environment - can serve as anything from a cabinet emulator to a wah-wah to completely new special effects) which can be switched on or off independently. The parameters in each section are as follows:

Damage section controls



The "Damage" section is where you specify the basic sound mangling algorithm and make settings to inflict the desired amount of damage to the sound. There are ten basic algorithms to choose from, ranging from classic distortion effects to digital-sounding warping and modulation effects.

There are five controls in this section, with the following functions:

Parameter	Description
Damage button	This switches the Damage section on or off.
Damage Control knob	This controls the input gain which in turn determines the amount of damage inflicted. The higher the value, the more destruction! When raising the Damage Control you may need to lower the Master level to maintain the same output level (and vice versa).
Damage Type knob	This selects the type of effect - see the table below for a description of the available damage methods.
P1/P2 knobs	The functionality of these knobs vary according to the selected Damage Type - see the table below for a description.

Description of the various Damage Type algorithms

Here follows a basic description of the ten Damage Types available, and what parameters the P1/P2 knobs control for each type:

Type	Description
Overdrive	This produces an analog-type overdrive effect. Overdrive is quite responsive to varying dynamics. Use lower Damage Control settings for more subtle “crunch” effects. <ul style="list-style-type: none">- The P1 knob controls the basic tone of the effect. Turn clockwise for a brighter sound.- The P2 knob controls Presence. Presence boosts frequencies in the high midrange <i>before</i> the distortion stage which in turn affects the character of the distortion. Turn clockwise for more Presence boost.
Distortion	Similar to Overdrive, but produces denser, thicker distortion. The distortion is also more “even” across the Damage Control range compared to Overdrive. <ul style="list-style-type: none">- The P1/P2 knobs control Tone and Presence, respectively - see Overdrive for a description.
Fuzz	Fuzz produces a bright and distorted sound even at low Damage Control settings. <ul style="list-style-type: none">- The P1/P2 knobs control Tone and Presence, respectively - see Overdrive for a description.
Tube	This emulates tube distortion. <ul style="list-style-type: none">- The P1 knob controls Contour, which is somewhat like a high pass filter, changing the tone and character of the distortion.- The P2 knob controls Bias, which changes the “symmetry” of the tube distortion. Setting this to the minimum or maximum value will produce asymmetrical distortion (typical of a real-life tube amplifier), while a 12 o’clock setting will produce symmetrical distortion (odd harmonics only).
Tape	This emulates the soft clipping distortion produced by magnetic tape saturation and also adds compression which adds “punch” to the sound. <ul style="list-style-type: none">- The P1 knob controls Speed, which simulates tape running at different speeds. The higher the Speed setting the more of the original high frequency material in the signal. Turn clockwise for a brighter sound.- The P2 knob controls the amount of Compression. Turning the knob clockwise increases the compression ratio.

Type	Description
Feedback	This effect combines distortion in a feedback loop which can produce many interesting and sometimes unpredictable results. Feedback is basically when a sound source is fed back to itself. An open microphone picking up sound from a nearby loud-speaker that is also being used to amplify sound from the microphone will produce a feedback loop with the associated typical howling. For this effect the Damage Control knob controls the gain of the feedback loop. <ul style="list-style-type: none">- The P1 knob controls Size, which could be described as the “length” (i.e. the distance between the microphone and the loud-speaker in the above example) of the feedback loop.- The P2 knob controls Frequency, which for this effect determines which overtones will “howl”.
Modulate	Modulate first multiplies the signal with a filtered and compressed version of itself, and then adds distortion. This can produce resonant, ringing distortion effects. <ul style="list-style-type: none">- The P1 knob controls Ring, which is the resonance of the filter. Turn clockwise for more pronounced ringing effects.- The P2 knob controls Frequency, which is the filter frequency. Turn clockwise to raise the filter frequency which generally produces a sharper, more piercing effect.
Warp	Warp distorts and multiplies the incoming signal with itself. <ul style="list-style-type: none">- The P1 knob controls Sharpness. Lower values will produce a soft, compressed distortion, while higher values produces more harmonics and a sharper sound.- An effect of multiplying a signal with itself is that the fundamental pitch is removed from the signal, leaving overtones only. The P2 knob controls Bias - raise this to reintroduce the fundamental pitch in the sound.
Digital	Lo-fi anyone? This reduces the bit resolution and sample rate for raw and dirty sounds or for emulating vintage digital gear. <ul style="list-style-type: none">- The P1 knob controls (bit) Resolution. If the knob is turned fully to the right there is no bit reduction, fully the left the resolution is 1-bit.- The P2 knob controls the sample rate. If the knob is turned fully to the right the sample rate is not reduced; turning it to the left gradually reduces the sample rate.
Scream	Similar to Fuzz, but with a bandpass filter with high resonance and gain settings placed before the distortion stage. <ul style="list-style-type: none">- The P1 knob controls the basic tone of the effect. Turn clockwise for a brighter sound.- The P2 knob controls the filter frequency. The high resonance setting of the filter makes it suitable for wah-wah effects.

Cut section (EQ)



The sliders in the Cut section are tone controls, allowing you to cut or boost the level by up to 18dB in the low, mid and high frequency areas. The Cut section is activated with the Cut button above the sliders.

Move the slider from the middle upwards to boost the level, and from the middle downwards to cut the level of the corresponding frequency area.

Body section



The Body section is just what it says - it places the sound in a resonant “body”. Depending on the settings, the result can be similar to a speaker cabinet simulator, an auto-wah effect, or effects with no real-world counterpart. The section is based on 5 basic body types, which simulate how a sound is affected by different physical enclosures. The size and resonance of the Body types can be changed, and the section also features an envelope follower.

The Body parameters are as follows:

Parameter	Description
Body button	This switches the Body section on or off.
Body Type knob	This is used to select one of the five available Body types (A-E).
Body Reso knob	This simulates the resonance of the selected Body. Turning the knob clockwise gives a more resonant effect.
Body Scale	The Body Scale parameter could be said to control the “size” of the Body. Note that this is “inverted” - turning the knob clockwise reduces the emulated size.
Auto knob	Determines the amount of envelope follower effect on the Scale parameter - see below.

About the envelope follower

The Body section features an envelope follower for dynamic control of the Scale parameter. The envelope follower analyzes the amplitude of the incoming signal and changes the Scale setting accordingly - the louder the incoming signal the higher the value of the Scale parameter. The operating frequency (or size) range is set with the Scale parameter, and the envelope follower amount is set with the Auto parameter. A typical use for this is auto-wah effects - try Body Type “B” for a pronounced wah effect.

- On the back of the Scream 4 you will find an Auto CV output - this delivers the CV signal from the envelope follower, allowing you to dynamically control parameters in other devices. See [page 229](#) for an example.

About the Master level control



The Master level control should be used when you need to increase or decrease the output level, while retaining the basic character of the effect. It can also be used to balance the level between the distorted sound and the “clean” (unprocessed) sound if the effect is to be switched in and out in the mix.

If the output level is too high, turning down the Damage Control setting would lower the output, but it would also change the character of the distortion, as would changing eq or presence settings.

Simply lowering the mixer channel level (for the channel that Scream 4 is connected to) would also work of course, but this would also mean that the level difference between the unprocessed and processed sound would increase.

So if the clip indicator lights up on the Transport, or if the distorted sound is too loud compared to the normal sound, the solution is to lower the Master output level.

As pointed out elsewhere in the manual, audio out clipping (indicated by the red clip indicator lighting up on the Transport panel) can *only* happen in the Reason Hardware Interface. In other words, you never have to worry about levels passed internally from device to device. However, bear in mind that if you use high Master output settings (or a lot of boost in the Cut section) Scream 4 can quite easily cause audio out clipping - and that is most likely *not* a distortion effect you want!

CV inputs and outputs



On the back of the Scream 4 you will find CV inputs for controlling the following four parameters:

→ **Damage Control**

Use this for dynamically changing the amount of damage effect.

→ **P1**

The use for this depends on the selected Damage Type. For example, if the Feedback effect is selected, this will control the Size parameter - connect it to the CV Out on a Matrix or synth LFO for strange, flanger-like sweeps.

→ **P2**

The use for this depends on the selected Damage Type. For example, if the Scream effect is selected, this will control the Frequency parameter, producing a distorted wah wah sound.

→ **Scale**

Lets you control the Scale parameter in the Body section from another CV source, for wah wah-like effects, etc.

In addition, you find a CV output from the "Auto" (envelope follower) function in the Body section. By connecting this to a CV input for a parameter in another device, the level of the signal going into the Scream 4 will affect that parameter. See below for an example on how to use this.

Tips and tricks

Don't restrict yourself to using Scream 4 as a basic distortion stompbox, but try it in as many ways as possible - you may be surprised to find how often Scream 4 can add power, warmth and color to your sounds. Here are some examples:

Creating a heavy drum sound

Scream 4 is ideal for processing drums. Try connecting it as an insert effect to a Redrum device and experiment with the Damage Types and settings.

- For classic distorted drums, try the Tube, Tape or Distortion algorithms.
- The Scream algorithm is excellent for a really raw body or industrial drum sound.
- For more weird, synth-like effects, try the Modulation or Warp effects.

Remember that you don't have to route the whole drum kit through the Scream device - sometimes it may be better to route the individual outputs from the bass drum, snare and/or toms to a Spider Audio Merger (see [page 248](#)), connect the merged output of the Spider to the Scream 4 and route this to a separate channel in the Mixer. That way, hi-hats, cymbals and similar are unprocessed.

Warming up a mix with the Tape effect

If you find your mixes a bit sterile, the Tape algorithm is excellent for providing some warmth and gentle distortion:

1. **Create a Scream 4 device and connect it between the main outputs of the Mixer and the Audio Hardware device.**
2. **Set the Damage Type to Tape.**
Make sure the Cut and Body sections are turned off.
3. **Start with a low Damage Control setting and P1 (Speed) and P2 (Compression) at 12 o'clock.**
4. **Play back your mix and adjust the settings.**

Raise the Damage Control for more tape saturation distortion, adjust P1 to get the desired brightness and raise P2 if you want a more controlled, compressed sound. If you like, you could also activate the Cut section and use the three-band EQ to further adjust the sound.

Using the Body section as a sound enhancer/phaser/wah

Nothing stops you from using the Body section on its own, without Damage. Try this:

1. **Create a sampler device (e.g. an NN-19) and select an electric piano patch.**
2. **Select the sampler and create a Scream 4.**
It is added as an insert effect.
3. **Turn off the Damage section and instead activate the Body section.**

You will find that this adds a resonant character to the sound, which will make it more “alive” and help it stand out in a mix. You should experiment with the Body settings to find the character that suits you best. You could also activate the Cut section- if you e.g. find the sound too bassy, just lower the “Lo” slider a bit.

4. **Now flip the rack around and connect the CV out from the sampler's LFO to the Scale CV input on the Scream 4.**

As you can hear, the Scale is modulated by the LFO.

5. **Adjust the amount of Scale modulation with the pot next to the CV input on the back of the Scream 4, and the speed (and waveform) of the modulation in the LFO section on the sampler device.**

With this type of modulation setup, it's easy to get lush, dreamy phaser effects. For a more wah wah-like sound, select Body type B and raise the Reso and Scale settings.

Emulating vintage digital gear

The first generations of digital instruments (drum machines, synths and samplers) used 8 or 12 bit sampling and processing, at low sample rates. This “lo-fi” sound is still in high demand, e.g. in hip-hop and R'n'B. Try this:

1. **Connect a Scream 4 as an insert for a Redrum, with a suitable kit selected.**
2. **Set Damage Type to Digital and turn P1 and P2 fully right.**
3. **Play back and lower P1 (bit depth) and P2 (sample rate) to get the sound you want.**
You may also want to use the Cut section to emphasize or cut frequencies in the sound.

Creating a real dynamic wah effect with the envelope follower

As we have shown, you can get auto-wah-like effects with the Body section in Scream 4 (by using the Auto parameter). You could also use the ECF-42 envelope controlled filter and trigger this with a gate signal - this is after all a “real” filter and sounds even more like a wah effect. However, to get a “real” auto-wah effect that responds to the signal level, you need to combine both these devices:

1. **Create an instrument device that you want to process with an auto-wah.**
It should be velocity responsive so that the harder you play, the louder it sounds.
 2. **Create a Scream 4 device and an ECF-42 device.**
Both these should now be connected as insert effects to the instrument device.
 3. **Turn off all three sections in the Scream 4.**
This is of course a matter of taste - but here we will show how to use the envelope follower in Scream 4, not its sound destruction capabilities.
 4. **Flip the rack around and connect the Auto CV Output on the Scream to the Freq CV input on the ECF-42.**
 5. **Lower the pot next to the CV input a bit - the envelope follower is rather sensitive and you probably don't want the filter to open too much.**
You can adjust this later if needed.
 6. **On the ECF-42, select the BP 12 (bandpass) mode and set the Res setting rather high.**
 7. **Play the instrument device and adjust the Freq setting on the ECF-42 to taste.**
As you can hear, the harder (or the more notes) you play, the more the filter will open.
- ✪ If you find the auto-wah too responsive, you could add a compressor between the instrument device and the Scream 4 to even out the level differences a bit.
- ✪ The Spider CV Splitter and Merger (see [page 249](#)) can be used to invert and split the Auto CV output for even greater flexibility.

RV7000 Advanced Reverb



The RV7000 is a high quality reverb processor. It features nine different reverb and echo algorithms, ranging from rooms and halls to special effects. Since the RV7000 comes with a number of useful reverb presets, you could simply select one and tweak the most important parameters on the main panel - or you could use the Remote Programmer panel to fine-tune the reverb in great detail.

The RV7000 also contains an equalizer and a gate section. Both of these are for processing the actual reverb sound, making it possible to get virtually any kind of reverb character, including gated reverb.

About the Patch format

Like the Scream 4 device, the RV7000 features programmable effect presets. In the Factory Sound Bank will find a number of preset Patches which can be used as they are or provide you with a good starting point for further tweaking.

Patches use the Windows file extension "*.RV7". Loading and saving Patches is done in the same way as for instrument devices.

Connections

Typically you connect the RV7000 as a send effect, as this allows you to use it for processing several different mixer channels. However, it's also possible to use it as an insert effect - use the Dry/Wet control on the main panel to adjust the balance between the dry, unprocessed sound and the reverb. Note:

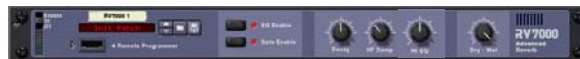
- ➔ **The RV7000 is a true stereo reverb, which means that it will use the stereo input information when processing both channels (without summing the input channels).**

It's also possible to use it as a mono in - stereo out effect. Which type of connection to use (mono or stereo in) depends on the material. If the audio sources are in mono (or in stereo but with no important difference between the left and right channel) using a mono input is sufficient.

- ➔ If you want to use RV7000's Reverse reverb effect, you should consider connecting it as an insert effect or using Send 4 on the Mixer, with Pre-fader mode selected (and the channel fader lowered).

This is because you typically don't want to hear the dry sound when using the Reverse effect. See [page 237](#).

The main panel



The RV7000 main panel.

When you create an RV7000, only the main panel will be shown. This contains a section for handling patches, on/off buttons for the EQ and Gate sections, the most important reverb parameters and a dry/wet mix control. To select a reverb patch and make coarse adjustments, this is all you need.

The remote programmer

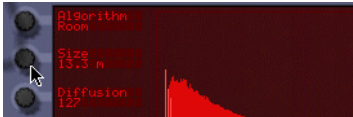
Clicking the arrow button next to the "cable slot" on the main panel brings up the remote programmer panel.



This is where you make detailed settings for the reverb. Note:

- The Edit Mode button to the left determines which section to make settings for, Reverb, EQ or Gate.

- Settings are made with the eight dials around the graphic display. The functions of the dials differ depending on the selected Edit Mode and the selected reverb algorithm. Next to each dial, the display shows the name and value of the corresponding parameter.



- Not all modes and algorithms use all eight dials. If a dial isn't used in the selected mode, nothing will be shown next to it in the display.
- You cannot make settings in the graphic display itself - this is for showing a graphic representation of the selected reverb.

Reverb algorithms and parameters

About the main panel parameters



On the main panel you find three reverb parameters that are available for all algorithms:

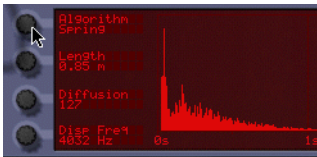
Parameter	Description
Decay	This governs the length of the reverb or the feedback if an echo algorithm is selected.
HF Damp	Controls how quickly the high frequencies should decay in the reverb. Raise it to gradually remove high frequencies, making the reverb sound warmer and less bright.
HI EQ	This is a high-shelving EQ that works much like a typical treble control on a mixer or amplifier. Lower the setting for a softer reverb sound or raise it to get more high frequencies.

Selecting an algorithm

You select a reverb algorithm in the remote programmer panel:

1. **Click the remote programmer arrow button on the main panel to display the remote programmer panel.**
2. **Make sure the Edit Mode button is set to Reverb.**
3. **Use the top left dial to select a reverb algorithm.**

The selected algorithm is shown in the display next to the dial.



Here's a quick overview of the nine algorithms - for details and parameter descriptions, see below.

Algorithm	Description
Small Space	Emulates a small enclosed space (a small room or a resonant body).
Room	Emulates a room with adjustable shape and wall character.
Hall	Emulates a hall.
Arena	Emulates a large arena, with separate pre-delay for the left, right and center reverbs.
Plate	Emulates a classic plate reverb.
Spring	Emulates a spring reverb, as used in e.g. guitar amplifiers.
Echo	An echo effect with gradually diffusing echo repeats. Can be synced to Reason's tempo.
Multi Tap	A multi-tap delay with four different delay lines and tempo sync.
Reverse	A reverse reverb effect that "pushes" the dry sound to appear after the reverb. The result is a backwards reverb leading up to the direct sound.

Small Space

This algorithm places the sound in a small enclosed space, ranging from a tiny resonant body to a room. The parameters are:

Parameter	Description
Size	The size of the emulated space.
Mod Rate	The reverb can be randomly modulated for a more even sound (or for special effects). This parameter sets the rate of modulation (the amount is set with Mod Amount).
Room Shape	Select from four different room shapes, affecting the character of the reverb.
LF Damp	Controls how quickly the low frequencies should decay in the reverb. Raise it to gradually remove low frequencies, making the reverb sound "thinner" and less boomy.
Wall Irreg	Adjusts the positioning of the emulated walls in the small space. The lowest setting emulates two directly opposed walls while higher settings emulate more walls and angles, for a more complex resonance.
Predelay	Sets the predelay time, i.e. the delay between the source signal and the start of the reverb.
Mod Amount	Sets how much the reverb will be modulated. Use fairly low settings when emulating real rooms and resonant bodies, and higher settings for special effects.

Room

Emulates a medium-sized room, with the following parameters:

Parameter	Description
Size	The size of the emulated room.
Diffusion	At low Diffusion settings, you will hear the individual reverb "bounces" more clearly, while higher settings produce a more "smeared", dense and even reverb.
Room Shape	Select from four different room shapes, affecting the character of the reverb.
ER->Late	The first "answers" in the reverb are called early reflections (ER) and are typically more pronounced than the actual reverb tail. This parameter sets the time between the early reflections and the reverb tail. This is set as a percentage - the actual delay time depends on the Size setting.
ER Level	Adjusts the level of the early reflections. "0" is normal level.
Predelay	Sets the predelay time, i.e. the delay between the source signal and the start of the early reflections and reverb.
Mod Amount	Sets how much the reverb will be modulated. Moderate modulation gives a natural, less static sound.

Hall

Emulates a hall. The parameters are the same as for the Room algorithm above (but the Hall algorithm offers larger Size settings).

Arena

Emulates the ambience in an arena or concert hall, with long pre-delay times (separate for left, right and center):

Parameter	Description
Size	The size of the emulated arena or hall.
Diffusion	At low Diffusion settings, you will hear the individual reverb "bounces" more clearly, while higher settings produce a more "smeared", dense and even reverb.
Left Delay	The predelay time for the left side of the reverb.
Right Delay	The predelay time for the right side of the reverb.
Stereo Level	Adjusts the level of the left and right sides of the reverb. "0" is normal level.
Mono Delay	The predelay time for the mono (center) reverb signal.
Mono Level	Adjusts the level of the mono (center) reverb signal. "0" is normal level.

Plate

A classic plate reverb, excellent for vocals for example. The parameters are:

Parameter	Description
LF Damp	Controls how quickly the low frequencies should decay in the reverb. Raise it to gradually remove low frequencies, making the reverb sound "thinner" and less boomy.
Predelay	Sets the predelay time, i.e. the delay between the source signal and the start of the reverb.

Spring

An emulation of a spring reverb as can be found in guitar amplifiers, organs, etc. The spring reverb has the following parameters:

Parameter	Description
Length	Sets the length of the simulated spring.
Diffusion	At low Diffusion settings, you will hear the individual reverb "bounces" more clearly, while higher settings produce a more "smeared", dense and even reverb.
Disp Freq	When sending a signal to a real-life spring reverb, the initial transient will produce a quick, characteristic sweeping tonal noise. This is because different frequencies in the sound are delayed by different amounts (a phenomenon called dispersion). This parameter controls the frequency of that sound.
LF Damp	Controls how quickly the low frequencies should decay in the reverb. Raise it to gradually remove low frequencies, making the reverb sound "thinner" and less boomy.
Stereo (on/off)	Determines whether the output of the spring reverb should be in mono or stereo.
Predelay	Sets the predelay time, i.e. the delay between the source signal and the start of the early reflections and reverb.
Disp Amount	Sets the amount of dispersion effect (see Disp Freq above).

Echo

This is an advanced echo effect, with diffusion controls and tempo sync. When Echo is selected, the Decay control on the main panel controls the echo feedback (the number of echo repeats). The parameters are:

Parameter	Description
Echo Time	Sets the time between each echo. When Tempo Sync (see below) is off, the echo time is set in milliseconds (10 - 2000 ms); when Tempo Sync is on you set the echo time as a number of 1/16 notes or 1/8 triplet notes, in relation to the current song tempo.
Diffusion	When this is set to 0, the echo will sound as a standard delay with clear, precise repeats. Raising the Diffusion setting will introduce additional echoes very close to the "main" echo repeats, causing a "smeared" echo sound. This will also expand the echo stereo image.
Tempo Sync	Determines whether the echo time should be freely set ("off") or synchronized to Reason's tempo ("on").
LF Damp	Controls how quickly the low frequencies should decay in the echoes. Raise it to gradually remove low frequencies.
Spread	Adjusts the spacing of the additional echoes added by the Diffusion parameter. For a very smeared echo (sound more like a reverb), set both Diffusion and Spread to their maximum values.
Predelay	Sets an additional delay time before the first echo repeat.

Multi Tap

The Multi Tap delay produces up to four different delays with separate delay times, panning and level. The whole set of four delay taps can then be repeated at a given rate. Again, the Decay control on the main panel controls the feedback (the number of repeats for the whole multi tap set). All delay times can be tempo synced.

Note: this algorithm is handled a bit differently since you make separate settings for each delay tap:

- The parameters to the left of the display are common for all taps.
- You use the Edit Select parameter in the top right corner to select which tap to make settings for - the three parameters below affect the currently selected tap.



Tap 2 selected for editing.

- You can also set Edit Select to "Repeat Tap" - this is where you specify the repeat time for the whole multi tap "package".
With short Repeat times, the first tap may be repeated before the last tap has sounded. This can be used to create very complex multiple delay effects.



The common parameters (to the left) are:

Parameter	Description
Tempo Sync	Determines whether the delay times and repeat times should be freely set ("off") or synchronized to Reason's tempo ("on").
Diffusion	Raising the Diffusion setting will introduce additional echoes very close to the "main" repeats, causing a "smeared" delay sound.
LF Damp	Controls how quickly the low frequencies should decay in the echoes. Raise it to gradually remove low frequencies.

When Tap 1 - 4 is selected with the Edit Select parameter, you can make the following settings for the selected delay tap:

Parameter	Description
Tap delay	Sets the delay - the time from the source signal to the tap. When Tempo Sync is off, the delay time is set in milliseconds (10 - 2000 ms); when Tempo Sync is on you set the delay as a number of 1/16 notes or 1/8 triplet notes, in relation to the current song tempo.
Tap level	Adjusts the level of the selected tap.
Tap pan	Adjusts the pan of the selected tap.

When Repeat Tap is selected with the Edit Select parameter, there is only one parameter to the right in the display:

Parameter	Description
Repeat Time	Sets the time between each repeat of the whole multi tap set. The number of repeats is set with the Decay control on the main panel. When Tempo Sync is off, the repeat time is set in milliseconds (10 - 2000 ms); when Tempo Sync is on you set the repeat time as a number of 1/16 notes or 1/8 triplet notes, in relation to the current song tempo.

Reverse

The Reverse reverb algorithm in RV7000 is special in that it actually "moves" the source audio as well. Sounds fed into the Reverse reverb are "sampled", a reverse reverb is created and played back and finally the "sampled" original sound is played back. For example, if you feed a snare drum hit into the Reverse reverb, you will hear a rising "backwards" reverb, followed by the snare drum hit.

Therefore, you probably don't want to hear the first, original (dry) sound. There are two ways to set this up:

- ➔ **Connect the RV7000 as an insert effect and make sure the Dry/Wet control on the main panel is set fully to "Wet".**
- ➔ **Connect the RV7000 as a send effect using send 4 on the Mixer, activate the Prefader (P) switch for the send and lower the mixer fader completely for the source signal.**
That way, the signal will be sent to the reverb but the dry sound from the Mixer channel isn't heard. Again, the Dry/Wet control on the reverb should be set to "Wet".

Note:

With this algorithm, raising the Decay setting on the main panel will make the reverse reverb start earlier and build up under a longer time. Similarly, the HF Damp parameter affects how fast the high frequencies are built up in the reverse reverb. In the remote panel, the Reverse algorithm has the following parameters:

Parameter	Description
Length	<p>This sets the time from when the source signal is fed into the reverb until it is played back again. It is during this time you will hear the reverse reverb, as shown in the display.</p> <p>The time can be set in milliseconds or as note values, depending on whether Tempo Sync is off or on.</p> <p>Note: As stated above, the Decay setting determines the length of the actual reverse reverb - in essence how soon it starts after the source signal. But of course, the reverse reverb cannot start <i>before</i> the original source signal! If you set Decay to a longer time than the Length setting, the reverse reverb will start abruptly, immediately when the source signal is fed into the reverb. If this sounds complicated, just take a look at the RV7000 display and try the settings - you will soon see how it works.</p> <p>Note also that very high Length settings demand a lot of processor power. This can be reduced by adjusting the Density parameter, see below.</p>
Density	<p>Density governs the "thickness" of the Reverse effect. If this parameter is turned down to zero, the effect produces individual delays rather than a dense "wash", which can be used as a special effect. Worth noting is that if Density is set to around 50%, this can considerably reduce the CPU load without altering the sound of the effect too much. Exactly how much the Density parameter can be reduced without altering the sound depends on the source material.</p>
Rev Dry/Wet	<p>Sets the balance between the "moved" source signal ("dry", low values) and the reverse reverb ("wet", high values).</p>
Tempo Sync	<p>Determines whether the Length setting should be freely set ("off") or synchronized to Reason's tempo ("on").</p>

The EQ section



The equalizer in RV7000 affects the wet reverb sound only and is used for shaping the character of the reverb. There are two EQ bands, one for low frequencies (shelving) and one full-range parametric EQ.

- ➔ **To activate the EQ, click the EQ Enable button on the main panel so that the indicator lights up.**
- ➔ **To make EQ settings, select “EQ” with the Edit Mode button to the left in the remote programmer panel.**
- ➔ **In this mode, the remote programmer display shows a frequency curve, indicating the settings you make with the EQ parameters.**

The parameters are:

Parameter	Description
Low Gain	The amount of cut or boost of the low-shelving filter.
Low Freq	The frequency below which the Low Gain cut or boost is applied.
Param Gain	The amount of cut or boost for the parametric EQ.
Param Freq	The center frequency of the parametric EQ, e.g. at which frequency the level should be decreased or increased.
Param Q	This governs the width of the affected area around the set center frequency. The higher the value, the narrower the affected frequency range.

- ➔ **Remember that you have a third EQ band at your disposal - the HI EQ parameter on the main panel.**
The reason why this is on the main panel and not in the EQ section is simply that it's a setting you may want to adjust often, without having to open the remote programmer panel.

The Gate section



The Gate section allows you to create gated reverb effects with a lot of options and possibilities. You can either trigger the gate from the source audio signal or via MIDI or CV.

When triggering the gate from the source audio signal, it works like this:

- The gate “listens” to the source (dry) signal and opens whenever the signal reaches a certain threshold level.
- The reverb sound is sent through the gate - when the gate is closed you won't hear the reverb.
- When the source signal level drops below the threshold level, the gate closes after a time that depends on the Hold parameter and the level of the source signal (see the parameter table).

- ➔ **If you need the gate to be open for an exact duration (time), you should trigger it via MIDI or CV.**
In audio trigger mode, the actual gate time will vary depending on the source signal.

When triggering the gate via MIDI or CV, it works like this:

- The reverb sound is sent through the gate - when the gate is closed you won't hear the reverb.
- Whenever the gate receives any MIDI note (sent to the RV7000) or a gate signal (connected to the Gate Trig CV input on the back of the RV7000), the gate opens for the duration of the note or gate signal.

Note:

- ➔ **To activate the Gate, click the Gate Enable button on the main panel so that the indicator lights up.**
- ➔ **To make Gate settings, select “Gate” with the Edit Mode button to the left in the remote programmer panel.**
- ➔ **In this mode, the remote programmer display shows two meters - one showing the signal level (with an indication of the threshold level) and one showing the status of the gate.**
This is useful for checking what happens, how the gate triggers, etc.

The parameters for the Gate section are:

Parameter	Description
Threshold	When Trig Source is set to "Audio", this determines the audio signal level at which the gate opens. If you raise this setting, only very loud sounds will open the gate.
Decay Mod	This modulates the reverb Decay parameter so that the decay time is lowered when the gate closes. When this is set to zero, no decay modulation happens - this means that if the gate is closed and then opened again, you may hear "previous" reverb tails that are still ringing. If you raise the Decay Mod setting, the decay will automatically be lowered when the gate is closed, eliminating this effect.
Trig Source	Determines whether the gate should be triggered by audio or MIDI/CV, as described above.
High Pass	A high-pass filter that affects the audio that triggers the gate (only active when Trig Source is set to "Audio"). If you raise this setting, sounds with low frequencies only will not open the gate. Note that this setting doesn't affect the sound of the reverb, only the triggering mechanism.
Attack	Determines how long it takes for the gate to open after a triggering signal has been received.
Hold	This parameter is only active when Trig Source is set to "Audio". Hold affects how quickly the gate closes, in the following way: Internally, the gate is controlled by an envelope follower that analyzes the source signal level and generates a "level CV signal" accordingly. This signal is compared to the Threshold level to determine whether the gate should be opened or closed. The Hold parameter affects how quickly the envelope follower responds when the source signal level drops - you could say that this is the decay control for the envelope follower. The higher the Hold setting, the longer it will take for the envelope follower signal to drop below the threshold level and close the gate. But the resulting time also depends on the source signal level - with a loud signal, it will take longer time for the envelope follower to drop to the threshold level. Therefore, the actual gate time depends both on the Hold setting and on the character of the source audio.
Release	Determines how long it takes for the gate to close after the Hold time.

CV Inputs



On the back of the RV7000 you find three CV inputs. These are:

Parameter	Description
Decay	Controls the reverb decay or echo/delay feedback via CV.
HF Damp	Controls the HF Damp parameter on the main panel.
Gate Trig	Used for triggering the Gate section with a gate signal. The length of the gate signal determines the length of the gated reverb.

RV-7 Digital Reverb



Reverb adds ambience and creates a space effect. Normally, reverb simulates some kind of acoustic environment such as a room or a hall, but you could also use it as a special effect.

- ➔ **The Reverb device can be used as a send effect or an insert effect.**
If several devices uses the same type of reverb, you should connect the reverb as a send effect, to conserve computer power.

Parameters

The display to the left on the panel shows the selected reverb algorithm - the general type of reverb. By clicking the arrow buttons you can change algorithm, with the following options available:

Algorithm	Description
Hall	Emulates a fairly large, smooth hall.
Large Hall	Emulates a larger hall, with pronounced pre-delay.
Hall 2	A hall reverb with a brighter attack than "Hall".
Large Room	Emulates a large room with hard early reflections.
Medium Room	Emulates a medium-sized room with semi-hard walls.
Small Room	A smaller room, suitable for "drum booth"-type reverbs.
Gated	A gated reverb, that is abruptly cut off.
Low Density	A thinly spaced reverb, where you clearly can here the individual echoes. Useful for strings and pads and as a special effect.
Stereo Echoes	An echo effect with the repeats alternating between stereo sides.
Pan Room	This is slightly similar to "Stereo Echoes", but the echo repeats have soft attacks.

- ⚙️ **If you need to conserve computer power, try using the Low Density algorithm. This uses much less power than the other algorithms.**

The selected reverb algorithm can be tweaked using the parameters on the device panel:

Parameter	Description
Size	Adjusts the emulated room size. Middle position (value 0) is the default size for the selected algorithm. Lowering this parameter results in a closer and gradually more "canned" sound. Raising this parameter results in a more spacey sound, with longer pre-delay. For the "Stereo Echoes" and "Pan Room" algorithms, the Size parameter adjusts the delay time.
Decay	This governs the length of the reverb effect. Middle position is the default decay time for the selected algorithm. Note: Decay is not used for the "Gated" algorithm.
Damp	Raising the Damp value cuts off the high frequencies of the reverb, thereby creating a smoother, warmer effect.
Dry/Wet	If you are using the reverb as an insert effect, you use this parameter to adjust the balance between the unprocessed audio signal (dry) and the effect (wet). If the reverb is used as a send effect, this should be set all the way to wet only, since you can control the balance by using the AUX send controls in the Mixer.

CV Inputs

You can control the Decay parameter via the CV input on the back of the Reverb device.

DDL-1 Digital Delay Line



This is a mono delay (where the output can be panned in stereo) that can be synchronized to the song tempo. The delay can be used as a send effect or an insert effect.

Parameters

Parameter	Description
Delay time	The display to the left on the device panel shows the delay time, either as note value steps (based on the sequencer tempo and the Step Length parameter) or in milliseconds, depending on the setting of the Unit switch. The maximum delay time is two seconds (2000 ms) while the maximum number of steps is 16. Note that if the tempo is low, you may reach the maximum delay time at a lower number of steps than 16 (in which case raising the steps value will not make any difference).
Unit	This is where you select whether you want a tempo-based delay ("Steps" mode) or a free time delay ("MS" mode). In the Steps mode, you specify the delay time in note value-based steps. This means that if you change the tempo in the transport panel, the delay will maintain its rhythmic relation to the music (provided that the resulting delay time doesn't reach the maximum value). This mode is useful for creating rhythmic patterns. If you change the tempo when using the delay in MS mode, the delay time will remain the same. See also the note about switching Unit modes below.
Step length	Governs whether each step in Steps mode should be a sixteenth note (1/16) or an eighth triplet note (1/8T).
Feedback	Determines the number of delay repeats.
Pan	Pans the delay effect to the left or to the right.

Parameter	Description
Wet/Dry	If you are using the delay as an insert effect, you use this parameter to adjust the balance between the unprocessed audio signal (dry) and the delay effect (wet). If the delay is used as a send effect, this should be set all the way to wet only, since you can control the balance by using the AUX send controls in the Mixer.

CV Inputs

The following CV inputs are available on the back panel of the device:

- **Pan CV.**
This allows you to control the panning of the delay signal. Connect an LFO to this for moving delay effects, or use a Matrix pattern to simulate random delay panning.
- **Feedback CV.**
This allows you to control the amount of feedback (the number of delay repeats) from another device. Useful for dub-type echoes on certain beats or notes only.

Switching between Unit modes

When you switch between the two Unit modes (Steps and MS), the following rules apply:

- **If you switch from Steps mode to MS mode, the delay will be set to the same actual delay time as was used in the Steps mode.**
This means that you can set up an exact rhythmic delay in Steps mode, and then switch to MS mode to adjust it slightly.
- **If you switch from MS mode to Steps mode, the delay is reset to the previously used Steps value.**

D-11 Foldback Distortion



The D-11 is a simple but effective distortion effect, capable of producing anything from just a whisper soft touch of distortion, to complete thrashing. This effect is most often used as an insert effect.

Parameters

The distortion has the following parameters:

Parameter	Description
Amount	This controls the amount of distortion. The higher the value, the more distortion.
Foldback	This adjusts the character of the distortion by introducing foldback, which makes the waveform more complex. The default value is in the middle position. This produces a "flat" clipping distortion, which is the most common type. Lowering the parameter makes the sound rounder and more gentle, raising it makes the sound sharper and more evil.

CV Inputs

On the D-11 you will find a CV input for controlling the Amount parameter. This can produce very drastic effects, especially if you control parameters in the instrument device (such as filter frequency and resonance) at the same time.

ECF-42 Envelope Controlled Filter



The ECF-42 is a multimode filter with a built in envelope generator. It is mainly designed to be used together with pattern devices to create pattern controlled filter and envelope effects, but it can also be triggered via MIDI, or used as a "static" filter for shaping the sound of an instrument device or a whole mix.

Usage

The Envelope Controlled Filter is best connected as an insert effect. However, unlike the other effects it is not a pure "stand-alone" device. To make the most of the ECF-42, you need either CV/Gate from an external device or MIDI notes from a sequencer track.

- ➔ **If you connect a device to the ECF-42 using audio inputs/outputs only, it will simply act as a filter with no velocity or envelope modulation.**
Hence, all filter parameters are "static", unless you manually turn the knobs or automate them in the sequencer.
- ➔ **Connecting a gate signal to the Env Gate input on the back panel of the device allows you to trigger the envelope generator for the filter.**
Note that the ECF-42 envelope generator is not triggered by the audio itself - the envelope parameters won't do anything unless the device receives gate signals.
- ➔ **By creating a sequencer track connected to the ECF-42, you can have the envelope triggered by MIDI notes on the track.**
The envelope is affected by the position, length and velocity of the MIDI notes (but not by their pitch).
- ★ **If you are unfamiliar with basic filter and envelope parameters, please refer to the Subtractor chapter for a description of these.**

The Filter Parameters



The ECF-42 filter section has the following parameters:

Parameter	Description
Mode	This button sets the desired filter mode. Three modes are available: 24dB/octave lowpass, 12dB/octave lowpass and 12dB/octave bandpass.
Freq	This is the filter cutoff frequency. When using the ECF-42 in "static" mode (without triggering the envelope), this parameter adjusts the frequency content of the sound. When using the envelope, the Freq parameter serves as the start and end frequency for the filter sweep.
Res	This is the filter resonance. Raising this produces a more extreme, "synthy" effect.
Env Amt	Determines how much the filter frequency should be affected when the envelope is triggered. The higher the value, the more drastic the effect. Note though, that if the Freq parameter is set high, raising the Envelope Amount will not make any difference over a certain value! This is because the filter is already fully opened - try lowering the Freq parameter in that case.
Velocity	This parameter determines how much the gate velocity value should affect the envelope amount.

The Envelope Parameters



This is a standard envelope generator with Attack, Decay, Sustain and Release parameters. It is triggered by a gate signal connected to the Env Gate input on the back panel, or by MIDI notes on a sequencer track connected to the ECF-42. The parameters have the following functionality:

Parameter	Description
A (Attack)	When the envelope is triggered, this is the time it takes before the envelope signal reaches its max value.
D (Decay)	After reaching its max value, this is the time it takes for the envelope signal to reach the sustain level.
S (Sustain)	If the gate remains open (or the MIDI note is held), the envelope signal will remain on this level.
R (Release)	When the gate is closed (gate CV goes back to 0) or the MIDI note ends, this is the time it takes for the envelope signal to drop from its current value to the start value (set by the Freq parameter).

→ **The Gate indicator lights up when the device receives a signal to the Env. Gate input on the back panel or a MIDI note from a sequencer track.**

CV/Gate Inputs

On the back panel of the ECF-42, you can find the following CV/Gate inputs:

- **Freq CV.**
Use this for controlling the filter frequency from another device. For smooth filter modulation, try connecting an LFO to this input.
- **Decay CV.**
For controlling the envelope decay parameter from another device.
- **Res CV.**
Allows you to control the filter resonance from another device. Can be very effective in combination with filter frequency sweeps.
- **Env. Gate.**
This is where you connect a gate signal (e.g. from a Matrix or Redrum device) for triggering the envelope.

Pattern Controlled Filter - An Example

This example shows how to use the ECF-42 and the Matrix to create pattern controlled filter effects. Proceed as follows:

1. **Start with an empty Song.**

2. **Create a Mixer.**

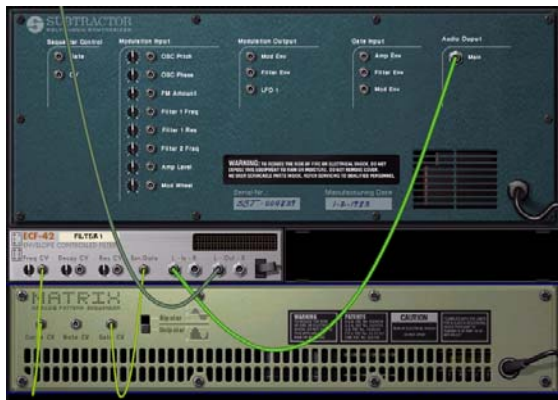
3. **Create a Subtractor Synthesizer.**

An Init Patch will work fine for these examples.

4. **Create an ECF-42.**

5. **Create a Matrix Pattern Sequencer.**

If you flip the rack around, you can see that the audio out from the Subtractor is passed through the ECF-42 and then on to the Mixer. The Matrix Curve CV is connected to the ECF-42 Frequency CV parameter, and the Matrix Gate CV is connected to the ECF-42 Env Gate input.



6. **Select the Track connected to the Subtractor (given that you are handling MIDI input via the sequencer) so that you can play it from your keyboard.**

If you play a few notes and turn the ECF-42 filter freq knob, you should hear the sound being filtered.

7. **Draw a Gate pattern in the Matrix, using mixed velocity values.**

Draw only a Gate pattern, not a Curve pattern.

8. **Set both the Env.Amt and Vel knobs on the ECF-42 to about “40”.**

9. **Click the Run button on the Matrix panel.**

10. **While in Run mode, hold a chord down on your keyboard.**

Now you should hear the envelope (controlling the filter) being triggered with every gate step.

→ **By increasing the Env.Amount, you determine how much the envelope parameters should affect the filter frequency.**

→ **By increasing the Vel. parameter, you determine how much the gate velocity should affect the filter frequency.**

✪ **If the filter effect isn't very noticeable, try lowering the filter frequency, and raising the Res value.**

11. **Set both the Env.Amt and Vel knobs on the ECF-42 to “0”.**

12. **With the Matrix still playing, draw a Curve pattern in the Matrix pattern window.**

Now, you should hear the filter frequency being modulated by the curve pattern. By combining the various parameters you can create many new filter effects.

→ **You can also control the ECF-42 from other devices with CV and/or Gate outputs.**

Triggering the ECF-42 via MIDI

To trigger the envelope in the ECF-42, proceed as follows:

1. **Create a sequencer track for the ECF-42.**

This is easiest done by bringing up the context menu for the device and selecting “Create Sequencer Track for XX” (where “XX” is the name of this particular filter device).

2. **Record or draw some notes on the sequencer track.**

Remember that the envelope takes the note length and velocity into account. The note pitches doesn't matter.

3. **Play back the track.**

The actual notes will not be heard (since the track is connected to the ECF-42, which produces no sound in itself) but the envelope will be triggered according to the notes.

→ **You can even control the envelope “live” via MIDI: just set MIDI input to the sequencer track for the ECF-42 and play your MIDI instrument!**

To route MIDI input to a track, click in the In column in the track list, so that the MIDI connector symbol appears next to the track name.

CF-101 Chorus/Flanger



The CF-101 is a combined chorus and flanger effect. It adds depth and movement to the sound by adding a short modulated delay to the audio signal. The delayed signal is then mixed with the original (either in the effect device or manually by you - see below). The CF-101 can be used as an insert or send effect.

Parameters

Parameter	Description
Delay	This is a manual control for the delay time used to create the chorus/flanger effect. Usually, flanger-type effects use fairly short delay times while chorus-type effects use medium long delays.
Feedback	This governs the amount of effect signal fed back to the input, which in turn affects the intensity and character of the effect. Turning this to the extreme left (negative feedback) or right (positive feedback) produces different flanger effects with a pronounced resonance "tone", while keeping it in between produces a more gentle chorus effect.
LFO Rate	This is the frequency of the LFO modulating the delay time. The higher the value, the faster the sound will oscillate.
LFO Sync	This button lets you activate/deactivate LFO sync. When it is activated, the frequency of the LFO is synchronized to the song tempo, in one of 16 possible time divisions. The LFO Rate knob is then used for setting the desired time division. Turn the knob and check the tooltip for an indication of the time division.
LFO Mod Amount	This determines the depth of the LFO modulation, i.e. by how much the delay time should be modulated. If you set this to 0, the effect will be "frozen" (most effective if you add some feedback).
Send Mode	This determines whether the delayed signal and the dry signal should be mixed in the effect device or not. If you use CF-101 as an insert effect, you should turn this off - the device will then output a mix of the dry signal and the modulated delay signal. If you use the device as a send effect, you should activate Send mode. Then, the device will only output the modulated delay signal, allowing you to mix it with the dry signal using the AUX send controls in the mixer. See also the note below about using the CF-101 as a vibrato effect!

CV Inputs

The following CV inputs are available on the back panel of the device:

- **Delay CV.**
Allows you to control the delay time from another device. This may give best results if you turn off the LFO modulation in the device (turn LFO Mod Amount to zero). For example, by controlling the delay parameter from a Matrix, you can create "stepped flanger" effects, in sync with the tempo.
- ★ **If you use the Delay CV input for "playing" the feedback tone, note that a higher delay value gives a lower pitch.**
- **Rate CV.**
Lets you control the rate of the modulating LFO from another device.

About Stereo and Mono connections

The CF-101 can be connected using mono or stereo inputs, as shown by the graphs on the back panel. Note the following:

- **A "moving" stereo effect is only obtained when you use a mono input and stereo outputs.**
With a stereo input, the two sides are processed in parallel, maintaining the stereo image of the original sound.
- **When you are using a mono input and stereo outputs, there will only be a stereo effect if the internal LFO is used.**
If you turn LFO Mod Amount to zero, both stereo outputs will carry the same signal (mono). This is because the "fake stereo" effect is produced by inverting the modulation for one of the output channels.

Tip: Using the CF-101 as a vibrato effect

The Send mode is intended for when using the CF-101 as a send effect. In this mode, the device will only output the modulated delay signal - you get the actual "chorusing" by mixing this signal with the dry, unprocessed signal in the Mixer.

However, if you activate Send mode while using the device as an insert effect, the result will be a pitch modulated version of the original sound - in short, a vibrato effect. Along with a little feedback, this can be used for special effects.

PH-90 Phaser



The PH-90 Phaser is a classic phaser effect with some special features for fine-tuning the sound. It can create the classic sweeping phaser sounds suitable for pads or guitars, but also more extreme effects if you like. The phaser is best used as an insert effect.

Theory

A phaser works by shifting portions of the audio signal out of phase, and then adding the processed signal back to the original one. This way, narrow bands of the frequency range ("notches") are filtered out. When these frequencies are adjusted, a sweeping phaser sound is created.

The PH-90 is a four-stage phaser, which means that there are four "notches" in the frequency response curve (this is a little like using four notch filters with different filter frequencies - see [page 108](#) for an explanation of notch filters).

When the phaser frequency is adjusted (manually or by the built-in LFO), these notches will move in parallel in the frequency spectrum. Furthermore, you can adjust the distance between the notches (Split) and their Width. Adding feedback raises the filter gain just below each notch in the frequency range, creating a more pronounced effect.

Parameters

Parameter	Description
Frequency	Sets the frequency of the first notch. Adjusting this will move the other notches correspondingly. This is the parameter modulated by the LFO to create phaser sweeps.
Split	This adjusts the distance between the notches in the frequency range, thereby changing the character of the effect.
Width	Determines the width of the notches. Raising the Width deepens the effect and simultaneously makes the sound more hollow and thin. This will also have an effect on character of the feedback "tone".
LFO Rate	This is the speed of the LFO modulating the frequency parameter. The higher the value, the faster the phaser sweeps.

Parameter	Description
LFO Sync	This button lets you activate/deactivate LFO sync. When it is activated, the frequency of the LFO is synchronized to the song tempo, in one of 16 possible time divisions. The LFO Rate knob is then used for setting the desired time division. Turn the knob and observe the tooltip that appears for an indication of the time division.
LFO Freq. Mod	This determines the depth of the LFO modulation, i.e. by how much the frequency parameter should be modulated. If you turn this to zero, the effect will be a static, formant-like sound (most effective if you add a little feedback).
Feedback	This is similar to the resonance control on a filter. Raising the feedback gives a more pronounced "tone" in the effect. For "singing" phaser sounds, try raising this to the maximum.

CV Inputs

The following CV inputs are available on the back panel of the device:

- ➔ **Freq CV.**
Adjusts the frequency parameter. Use this e.g. for creating envelope controlled phasing (preferably with LFO Freq. Mod turned off in the device).
- ➔ **Rate CV.**
Lets you control the speed of the modulating LFO from another device.

About Stereo and Mono connections

The PH-90 can be connected using mono or stereo inputs, as shown by the graphs on the back panel. Note the following:

- ➔ **A "moving" stereo effect is only obtained when you use a mono input and stereo outputs.**
With a stereo input, the two sides are processed in parallel, maintaining the stereo image of the original sound.
- ➔ **When you are using a mono input and stereo outputs, there will only be a stereo effect if the internal LFO is used.**
If you turn LFO Mod Amount to zero, both stereo outputs will carry the same signal (mono). This is because the "fake stereo" effect is produced by inverting the modulation for one of the output channels.

UN-16 Unison



The UN-16 simulates the sound of several detuned voices playing the same notes simultaneously. The voices are individually slightly delayed and also pitch modulated by low frequency noise. This produces a rich chorus effect with the voices spread across the stereo field (given that stereo outputs are used).

The UN-16 can be used as an insert effect or a send effect.

Parameters

Parameter	Description
Voice Count	This switch sets the number of voices for the effect; 4, 8 or 16.
Detune	This sets the amount of detuning for the voices. Turn clockwise for stronger detuning effects.
Dry/Wet	If you are using the UN-16 as an insert effect, you use this parameter to adjust the balance between the unprocessed audio signal (dry) and the effect (wet). If the UN-16 is used as a send effect, this should be set all the way to wet only, since you can control the balance by using the AUX send controls in the Mixer.

CV Input

One CV input is available on the back panel of the device. This controls the Detune parameter.

COMP-01 Auto Make-up Gain Compressor



The COMP-01 compressor levels out the audio, by making loud sounds softer. To compensate for the volume loss, the device has an automatic make-up gain, that raises the overall level by a suitable amount. The result is that the audio levels become more even and individual sounds can get more "power" and longer sustain.

The COMP-01 should be used as an insert effect, either for a single instrument device or for a whole mix (e.g. inserted between a Mixer device and the Hardware Interface). There are no CV inputs for this device.

Parameters

Parameter	Description
Ratio	This lets you specify the amount of gain reduction applied to the signals above the set threshold. The value is expressed as a ratio, from 1:1 (no reduction) to 16:1 (levels above the threshold are reduced by a factor 16).
Threshold	This is the threshold level above which the compression sets in. Signals with levels above the threshold will be affected, signals below it will not. In practice, this means that the lower the Threshold setting, the more the compressor effect.
Attack	This governs how quickly the compressor will apply its effect when signals rise above the set threshold. If you raise this value, the response will be slower, allowing more of the signal to pass through the compressor unaffected. Typically, this is used for preserving the attacks of the sounds.
Release	When the signal level drops below the set threshold, this determines how long it takes before the compressor lets the sound through unaffected. Set this to short values for intense, "pumping" compressor effects, or to longer values for a smoother change of the dynamics.
Gain meter	This shows the amount of gain reduction or increase (in dB), caused by the combined compression and make-up gain.

PEQ-2 Two Band Parametric EQ



While there is a simple two-band shelving equalizer available for each channel in the mixer, the PEQ-2 gives you much more precise control over the tone color. The device consists of two independent, fully parametric equalizers and is most often used as an insert effect, in mono or stereo.

About the two EQ modules

The two independent EQs are labeled “A” and “B”.

- ➔ **EQ A is always active (provided that the effect device is in “On” mode and that you have set the Gain to a value other than 0).**
- ➔ **To activate EQ B, click the button next to the EQ B parameters, so that the LED lights up.**
If you only use one EQ, it’s a good idea to turn EQ B off, to conserve computer power.

Parameters

For both EQs (A and B), the following parameters are available:

Parameter	Description
Frequency	This determines the center frequency of the EQ, e.g. at which frequency the level should be decreased or increased. The range is 31 Hz to 16 kHz.
Q	This governs the width of the affected area around the set center frequency. The higher the value, the narrower the affected frequency range.
Gain	Specifies how much the level of the selected frequency range should be boosted (positive values) or lowered (negative values). The gain range is ± 18 dB.

About the graphic display

The graphic display to the left in the device panel shows the frequency response curve as set by the EQ parameters. This gives a visual feedback and helps you tailor the EQ settings.

CV Inputs

The following CV inputs are available on the back panel of the device:

- ➔ **Freq 1 CV.**
Allows you to control the frequency of EQ A from another device, creating subtle or dramatic EQ sweeps depending on the Q and Gain settings.
- ➔ **Freq 2 CV.**
Allows you to control the frequency of EQ B in the same way.

Spider Audio Merger & Splitter



The Spider Audio Merger & Splitter is not an effect device, but a utility. It has two basic functions:

- To merge up to four audio input signals into one output.
- To split one audio input signal into four outputs.

There are no controls on the front panel of this device, only signal indicators.

Merging audio



Four audio input pairs. → Merged outputs.

On the back panel of the Spider are several audio connectors. The left half of the panel contains four stereo audio input connectors, and to the right of these, one merged stereo output.

- **The principle is simple; all audio signals connected to any of the four inputs will be merged and output via the output connectors.** If you connect a mono signal (to a L/Mono input, with nothing connected to the corresponding R input) it will be output on both merged outputs. This way you can merge stereo and mono signals freely. If you connect a signal to the R input only (with nothing connected to the corresponding L/Mono input) it will be output on the R output only.

Practical uses of merging audio

There are many practical uses of merging audio signals together, for example:

- **Process several audio signals with the same insert effect(s).** Perhaps you want to process certain channels in a mix with the same compressor, or use one ECF-42 to filter a group of instruments in a mix. You can also set up a chain of insert effects and process the merged signals.
- **Sub-grouping signals.** It may be practical to control several audio signals using one channel strip in the Mixer.
- **Use merged signals as either carrier or modulator source for the BV512 Vocoder.** You could use several sounds as carrier signal, or modulate the carrier with several signal sources.

Splitting audio

The right half of the back panel contains two signal splitters, labeled "A (L)" and "B (R)". The two splitters work independently, in the following way:

- **The signal fed to the input connector will be simultaneously output by all four outputs.** For splitting stereo signals you simply use both splitters with "A" for the left channel and "B" for the right channel.



Inputs. → Four split output pairs.

Practical uses of splitting audio

There are many practical uses of splitting audio signals - here a few examples:

- **Create "pseudo" stereo effects from mono signals.** For example, you could route the mono output of a Subtractor to the Spider and then send two split outputs (from the same row) to different effects and on to different Mixer channels panned left and right.
- **It provides a way to instantly switch between (or mix) different variations of the same signal.** This is a neat way of applying "spot effects" in a mix. An instrument output is split and sent to three different combinations of insert effect processing. The outputs from the three effects are routed to separate channels in the Mixer, which could in turn have different send effects, eq, etc. You then have three different variations of the same signal that can be easily switched in and out of the mix for drastic sonic changes - or combined for huge layered sounds.

Spider CV Merger & Splitter



The Spider CV Merger & Splitter is not an effect device, but a utility. It has two basic functions:

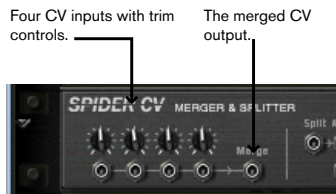
→ **To provide one merged CV output from up to four CV input sources.**

→ **To split CV or Gate inputs into several outputs.**

Two inputs, A and B, are provided, each with four outputs, where one of the outputs will invert the polarity of the control signal. One reason for having two splittable inputs is to make it possible to split Gate and Note CV, to connect several instrument devices with one Matrix for example.

There are no controls on the front panel of this device, only CV signal indicators. The four horizontal indicators light up to indicate signals connected to the corresponding merge input. The two indicators to the right indicate signals connected to the corresponding split inputs.

Merging CV



On the back panel of the Spider there are several CV connectors. The left half of the panel contains four CV/Gate input connectors with associated trimpots, and to the right of these, one merged CV output.

→ **The merged CV output will produce a CV signal that represents the “sum” of all connected CV inputs.**

A few things to note:

- Gate CV signals typically trigger notes or envelope cycles and are normally routed to a Gate input.
- CV signals typically control note pitch or for modulating parameters and are typically routed to CV Note or Modulation inputs.

There are no strict rules involved, but the facts mentioned above means that it is generally better to stick to using either Gate CV signals or CV signals but not a mixture when merging. simply because the CV/Gate signals usually go to different input destinations.

For instance, merging Note CV and Gate CV from a Matrix does not make much sense if you want to use Matrix to play melodic patterns via the Sequencer Control inputs of an instrument device. There would only be one merged output whereas the instrument device would need a separate Gate and Note CV signal to work properly.

Practical uses of merging CV

The practical applications of merging CV are maybe less obvious compared to splitting CV. But there are numerous applications for a merged CV control output, a few of which are listed below:

→ **You can create interesting modulation effects by merging several Modulation outputs from LFO's and other CV modulation sources.**

For example, merging the Modulation outputs from several LFO's would produce a “mixed modulation” output. This merged output signal could be likened to a “super LFO” capable of generating several modulation cycles simultaneously, each with a different waveform and modulation rate! In addition to this, by using the trim control for each CV input, you have full control over the *amount* of modulation applied by each LFO.

The above example could of course also include Curve CV outputs from a Matrix or Mod Outs from Malström etc., in short any CV Modulation output.

→ **Use the ECF-42 Filter to apply envelope controlled filter effects.**

This can create the sound of “synthesized” percussion, and other interesting effects.

This is done using the following method:

1. **Connect the audio outputs of a Redrum to a ECF-42 Envelope controlled filter.**
2. **Connect the Gate outputs from up to 4 Redrum drum channels to the merge inputs of a Spider CV.**
3. **Route the merged output to the Env Gate input on the ECF-42.**
If you add a touch of velocity the connected Gate signals will trigger the ECF-42 filter envelope. Again, the trim pots on the Spider allows you to adjust the amount of filter envelope applied.

→ **Create an “arpeggiator” using two Matrix devices and the Spider CV Merger & Splitter.**

By merging the Note CV output from one Matrix with a Curve CV output of another Matrix, you can transpose the Matrix pattern in real-time, a bit like an arpeggiator.

1. Create a Subtractor and a Matrix device.

Connect the Matrix Note and Gate CV outputs to the Subtractor Sequencer CV and Gate inputs, respectively.

2. Program a pattern for the Matrix.

In the following text this is referred to as “Matrix 1”

3. Now create a Spider CV and a second Matrix device and connect them as in the picture below.



Note that the Note CV output from Matrix 1, and the Curve CV output of Matrix 2 should be connected to the Spider. The merged output is connected to the Sequencer Control Note CV input on the Subtractor.

4. On the Spider CV, turn the trimpot for the input connected to the Note CV output fully to the right.

This setting will retain the correct pitch relationship for the notes played by the pattern.

5. On the Spider CV, turn the trimpot for the input connected to the Curve CV output to “32”.

This will produce a Curve CV output that corresponds to semitone steps.

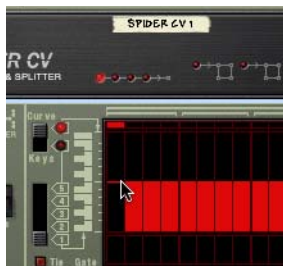


6. Set the Curve type switch to “Bipolar” on the back of the second Matrix (Matrix 2).

7. Flip the rack around so that the front panels are showing, and make the following settings for the “Matrix 2”:

- Set the number of steps to “1”.
- Set the Curve/Keys switch to “Curve”.

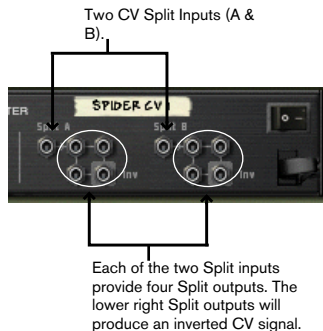
8. Adjust the Matrix 2 curve for step 1 (the only step used) so that it is in the middle of the bipolar curve as the picture shows.



9. If you now activate Play from the transport, the pattern you programmed for Matrix 1 is played back. By carefully adjusting the Matrix 2 Curve step 1 up or down the Matrix 1 pattern is transposed in semitone steps.

By programming different values for the "pattern" played by Matrix 2 and saving them in different pattern locations, you can use the Pattern selectors to transpose the Matrix 1 pattern to different keys!

Splitting CV



On the right half of the back panel you will find two split inputs "A" and "B", each with four output connectors. The signal connected to a Split input will be output by all four corresponding outputs, where one is inverted.

Practical uses of splitting CV

There are many practical uses of splitting CV signals - here are a few examples:

- **Connecting the CV Note and CV Gate outputs from a Matrix to Split Input A and B, allows you to connect the Matrix to several instrument devices.**

Simply route the CV and Gate outputs to the corresponding Sequencer Control CV/Gate inputs on the instrument devices. Although this could also be done by copying the Matrix Pattern data to several sequencer tracks and routing the outputs to the desired devices, the advantage by using Split is if you are editing Matrix pattern data this will be immediately be reflected in all the connected devices, without any copy/paste operations.

- **Splitting modulation outputs from LFO's, Curve CV data etc. allows you to apply modulation from one source to several parameters.**

By using the inverted output, you can create interesting modulation cross-fades, where one parameter value rises and another parameter value is lowered for example.



REASON

21

→ Menu and Dialog Reference

Reason Menu (Mac OS X)

About Reason

This menu item opens up a dialog that informs you about the version of the program and the people behind it.

Preferences

This menu item opens up the Preferences dialog. See [page 268](#) for detailed descriptions of the options in this dialog.

In addition, the Reason menu contains the standard Mac OS X services and Hide/Show options. Please consult the Macintosh help for descriptions of these options.

Quit Reason

This allows you to quit the program. If there are any documents open with unsaved changes you will be asked whether you want to save those changes.

File Menu

New

When you select this, a new, empty song appears. The exact contents of this song is determined by your Preferences settings (see [page 268](#)).

Open...

To open a Song, proceed as follows:

- 1. Pull down the File menu and select Open.**
The Reason song browser window appears.
- 2. Use the browser to navigate to the desired folder on disk or within a ReFill.**
- 3. When you have located the song file, select it and click Open (or double click on the file).**
The song appears in its own document window.

- ★ You can have several songs open at the same time if you like. This allows you to copy and paste patterns and patches between songs. However, all open songs consume some memory and performance, so you may want to close songs you don't need.

Close

This closes the active window.

If the window is a song document and it has unsaved changes, you will be asked whether you want to save those changes.

Save

This saves the active song document to disk.

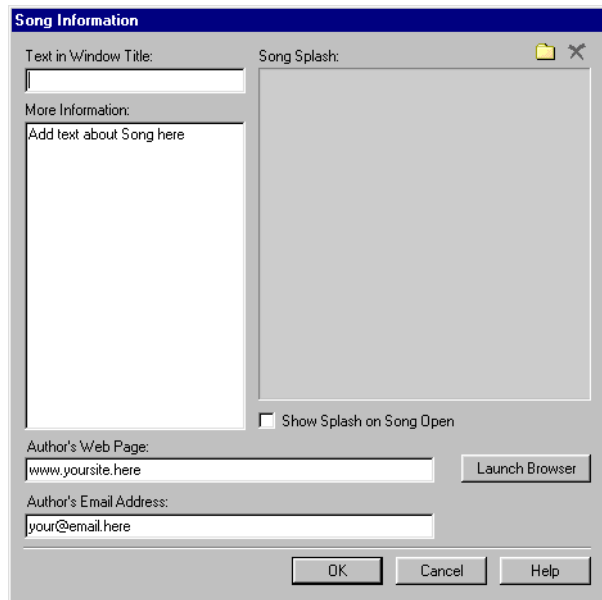
- If the song document hasn't yet been saved, the Save As dialog appears, requesting you to enter a file name and specify a location for the file on disk.
- If the document has already been saved at least once, the document will simply be saved without any questions.

Save As...

This saves the active song document to disc. A standard Save As dialog appears requesting you to enter a file name and specify a location for the file on disk.

- ★ You can set things up so that any samples used in the song are included in the song file itself by specifying self-contained settings (also on the File menu).

Song Information...



This brings up a dialog that allows you to add contact information, comments about the song, etc. Furthermore, if you save a published version of the song in the Reason Song Archive on the Propellerhead web site, vital information can automatically be extracted by the web archive engine, and displayed with the song file.

The dialog contains the following items:

Text in Window Title

The text you add here will be displayed directly after the file name in the song window's title bar.

More Information

This is where you add notes and comments about the song.

Song Splash

Allows you to add a picture to the song. If the "Show splash on song open" checkbox is ticked, the picture will be displayed when the song is opened.

To add a splash picture, click the folder button at the upper right corner, and locate and open the picture file in the file dialog that appears.

! Splash pictures must be JPEG files (Windows extension ".jpg") with a size of 256 x 256 pixels.

To remove the splash picture from the song, click the cross button.

Author's Web Page

Allows you to specify your web site. The user can go directly to your site by clicking the Browser button to the right (provided there is a working Internet connection).

Author's Email

This is where you specify your E-mail address, if you want other Reason users to send you their comments, etc.

Publish Song...

If you want to make your songs available to the public, e.g. for downloading on the Internet, there is a special file format for this. A Reason published song (Windows file extension ".rps") is much like a self-contained song, but has the following restrictions:

- The user cannot save any changes to the song.
- Copy, Cut and Paste is disabled.
- It is not possible to use the function Export Song/Loop as Audio File.

In a word, published songs are "locked". They are meant for playback only - no elements can be added, removed or extracted. Furthermore, a published song contains information about which ReFills are required (if any).

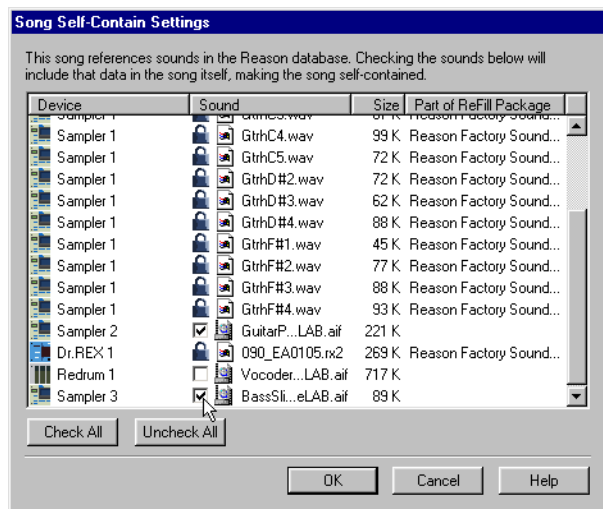
To create a published song, pull down the File menu and select Publish Song. Specify a name and location for the published song in the file dialog that appears, and click Save.

→ **Note that you don't have to make self-contained settings - all files (except ReFill components) are automatically included.**

About the Reason Song Archive

On the Propellerhead web site (www.propellerheads.se) you will find the Reason Song Archive. This allows you to share your music with other Reason users by uploading your songs.

Song Self-contain Settings...



A self-contained song contains not only the references to the used files, but also the files themselves. You can choose exactly which files should be included in the self-contained song, with the following exception:

! Files that are part of a ReFill cannot be included in a self-contained song.

If your song contains samples or REX files from a ReFill, other users must have the same ReFill to be able to play the song.

To specify which files should be included in the song, proceed as follows:

- 1. Tick the checkbox in the Sound column for the files you want included in the song.**
 - You can use the **Check All** button to activate all checkboxes in one go. Similarly, the **Uncheck All** button deactivates all checkboxes.
 - **Files that are part of a ReFill are indicated by a lock symbol instead of a checkbox (since they cannot be included in the song file).** The rightmost column indicates to which ReFill each such file belongs.

- 2. When you have selected the desired sounds, click OK.**

The dialog is closed. The next time you save, the specified sounds will be included in the song file.

- ! Note that a self-contained song file will be considerably larger than the original song file. However, samples included in a self-contained song are automatically compressed by approximately 50%, meaning that the self-contained song will still be a lot smaller than the original song and the sample files combined.**

“Un-self-containing” a Song

If you have opened a song that is more or less self-contained (i.e. contains one or several sounds embedded in the song file), you may want to extract these sounds and make the song refer to them on disk as usual.

- 1. Locate the sounds you want to extract from the song file, and deactivate their checkboxes (or click Uncheck All).**

- 2. Click OK to close the dialog.**

Now, the program will check for each “extracted” sound file whether it is available in your database (at its original, stored location) or not.

- If the program finds the sound file at the location stored in the song, it is simply removed from the song file, and the original file reference path is used.
- If the program doesn't find the sound file, a file dialog appears, allowing you to select a folder and name for the sound file.

Import MIDI File...

Reason can import standard MIDI files (SMF). This allows you to import MIDI data to Reason from other applications.

- ➔ **Under Windows, MIDI files have the extension “.mid”.**
On a Macintosh, MIDI files are recognized if they have the file type “Midi”.
- ➔ **If the imported MIDI file is of “Type 1”, there will be one sequencer track for each track in the MIDI file.**
- ➔ **If the imported MIDI file is of “Type 0” (that is, it contains one track with MIDI events on multiple channels), there will be one sequencer track for each used MIDI channel.**
- ➔ **Any tempo changes in the MIDI file are disregarded.**
The tempo in Reason will be set to the first tempo in the MIDI file.
- ➔ **The new tracks will not be connected to devices in the rack.**
You will need to connect the tracks manually to the proper devices, by using the Out pop-up menu in the track list.
- ➔ **All controller data in the MIDI file is included.**
This means that pitch bend, volume and modulation wheel data are preserved properly. However, some controllers may “mean” different things for the original MIDI instruments used when creating the MIDI file and the devices in Reason. When you have connected a sequencer track to a device, you may therefore need to remove some unwanted automation from the track.

Export MIDI File...

Reason can export standard MIDI files (SMF). This allows you to transfer MIDI data from Reason to other applications.

1. **Set the End (E) marker at where you want the MIDI file to end.**
The MIDI file will contain all events on all tracks from the start of the song to the End marker.
2. **Select “Export MIDI File” from the File menu.**
3. **In the file dialog that appears, specify a name and location for the file.**
Under Windows, the file will automatically get the extension “.mid”. Under Mac OS, this is not required. However, if you want the MIDI file to be recognizable under Windows (and by some hardware sequencers), you may want to activate the option “Add Extension to File Name” before saving.
4. **Click Save.**

MIDI files exported by Reason will have the following properties:

- ➔ **The MIDI file will be of Type 1, with one MIDI track for each track in the Reason sequencer.**
The tracks will have the same names as in the Reason sequencer.
- ➔ **Since the Reason sequencer doesn't use MIDI channels as such, all tracks will be set to MIDI channel 1.**
- ➔ **The sequencer tempo is included in the MIDI file.**

Export Device Patch...

This item is valid for all items that can save patches. The menu item name reflects the type of device selected (for example “export Redrum Patch”).

Even though the device settings are stored in the song, you may want to save any settings you have made for a device as a separate patch file. This allows you to use the patch in other songs, and lets you try out other patches in your song without risking to lose your sound.

- ➔ **Under Windows, the different types of patch files have different file extensions.**
These are “.zyp” (Subtractor patch files), “.smp” (NN-19 patch files) “.drp” (Redrum patch files), “.xwv” (Malström patch files) and “.sxt” (NN-Xt patch files). Under Windows, file extensions are automatically added by Reason when you save. Under Mac OS, you can choose to automatically add extensions by activating the “Add Extension to File Name” checkbox in the save dialog (this is not required, but may be a good idea if you want the saved files to be usable under Windows).
- ➔ **If you have selected a patch, modified it and want to save it with the modifications, you could either save a separate, modified version of the patch (with a new name) or simply overwrite the old patch file on disk.**
As usual, you will be asked whether you really want to replace the existing patch file.
- ★ **You can save a patch under the same name and location (without having the save dialog appear) by holding down [Option] (Mac) or [Alt] (Windows) and clicking the floppy disk button on the device panel. Note that this overwrites the original patch!**

Export Song/Loop as Audio File...

When you have created a complete song, you may want to mix it down to an audio file to make it playable for other people (who don't use Reason). You can either export the whole song (from the start to the “E” marker), or only the loop (the area between the left and right locator in the sequencer). Proceed as follows:

1. **Make sure only the main stereo outputs are used.**
The export function will only include audio routed to the stereo outputs.

2. Make sure the Loop/End markers are at the correct positions.

If you want to export the loop, you need to set the left and right locators to encompass the desired area. If you instead want to export the whole song, make sure the End (E) marker is at the desired end position.

★ If you are using reverb or delay, you may want to adjust the right locator or End marker so that the reverb/delay “tails” are included in the exported file.

3. Check that the song (or loop) plays back properly.

It's especially important that no clipping occurs during playback (see [page 74](#)).

4. Pull down the file menu and select Export Song as Audio File (or Export Loop as Audio File).

A file dialog appears.

5. Specify a name, location and file type (AIFF or Wave) for the audio file, and click Save.

6. Specify a sample rate and bit depth (resolution) for the exported file in the Settings dialog that appears.

7. Click OK.

The program creates the audio file. Depending on the length of the song/loop, this may take a while, during which a progress dialog is shown.

Export REX as MIDI File...

If you have imported a REX file into a Dr. Rex device and wish to play back the loop via MIDI (typically from another sequencer), proceed as follows:

1. Select the Dr. Rex device in the rack.
2. Select “Export REX as MIDI File...” from the File menu.
3. Save the MIDI File to disk.
4. In the other application, open the MIDI file you just created.
5. Set up the other application to play back the MIDI File on the correct MIDI Output and MIDI Channel (the output and channel on which the Dr. Rex device receives data).

Quit

This allows you to quit the program. If there are any documents open with unsaved changes you will be asked whether you want to save those changes.

Edit Menu

Undo

Virtually all actions in Reason can be undone. This includes creation, deletion and reordering of devices in the Rack, parameter value adjustments, editing in the sequencer and tempo/time signature adjustments. You can undo up to 10 actions.

→ To undo the latest action, select “Undo” from the Edit menu or hold [Command] (Mac) or [Ctrl] (Windows) and press [Z].

The action to be undone is indicated next to the Undo command on the Edit menu. For example, if your latest action was to delete some device(s) from the Rack, the Edit menu will say “Undo Delete Devices”.

Redo

To redo an undone action (“undo the undo operation”), select “Redo” from the Edit menu or hold [Command] (Mac) or [Ctrl] (Windows) and press [Y].

The action to be redone is indicated next to the Redo command on the Edit menu.

You can undo/redo up to 10 actions.

Cut/Cut Device/Cut Pattern

This command takes the selected item(s), removes them and places them on the clipboard (an invisible storage location) from where they can later be pasted in.

Cutting applies to tracks, sequencer events and Groups, devices and patterns.

Copy/Copy Device/Copy Patch/Copy Pattern

This command takes the selected item(s), copies them and places the copies on the clipboard (an invisible storage location) from where they can later be pasted in.

Copying applies to tracks, sequencer events and Groups, devices and patterns.

Paste/Paste Device/Paste Patch/Paste Pattern

This command takes the items you have cut or copied and pastes them back into the document.

Sequencer tracks

→ If you Paste the track(s) into their original song, this simply duplicates the tracks.

However, the Pasted tracks will not be connected to any devices in the rack.

→ **You can also Paste the track(s) into another song.**

Note that only the tracks (complete with contents) are copied and pasted - not their respective devices. You may want to separately copy and paste the devices to the other song.

Sequencer events and groups

→ **When you Paste sequencer events and groups, they appear at the song position, on their original track(s).**

If you have deleted the original tracks, or if you Paste into another Reason song document, new tracks will be created as needed.

Devices

→ **When you Paste devices, these are inserted into the rack below the currently selected device.**

If no device is selected, the pasted devices will appear at the bottom of the rack.

→ **If you Copy and Paste several devices, the connections between these are preserved.**

→ **If you hold down [Shift] when you Paste a device, Reason will attempt to automatically route its audio patching.**

Patterns

→ **Paste Pattern copies the pattern on the clipboard to the selected pattern location in the selected device.**

This overwrites the selected pattern with the one on the clipboard.

Transferring patterns between Reason songs

If you want to copy patterns between different Reason songs, you use copy and paste:

1. **Open both songs.**
2. **Select the pattern you want to copy.**
3. **Select Copy Pattern from the Edit menu or the device context menu.**
4. **Make the other song active.**
This is done by clicking in the song window or by selecting the song from the Windows menu.
5. **Select the bank and pattern location to which you want to copy the pattern.**

Note that any pattern already stored in that location will be overwritten!

6. **Select Paste Pattern from the Edit menu or the device context menu.**

Clear/Delete Device/Clear Pattern

This menu item is used for deleting selected items. It is also used for clearing (emptying) the select pattern on a pattern device.

Initialize Patch

Sometimes it is useful to start with a “clean slate” when creating a synth sound, a drum kit or a sampler patch. This is done by selecting Initialize Patch from the device context menu or Edit menu. This sets all parameters to “standard” values. Initializing NN-19, NN-XT, Dr. Rex or Redrum devices will also remove all sample file references, allowing you to start from scratch.

Select All

This selects all items of the same type as the one currently selected, that is, all sequencer notes, all devices in the rack, etc.

You can use this to quickly apply a command to all items you are working on, for example deleting all devices in the rack (select Select All and then press [Delete]) or for Quantizing all notes in the Edit View (select Select All and then click the Quantize button).

→ **Whether the Select All function applies to the track list, the Arrange/Edit View or the rack depends on which area has focus in the program.**

Focus is indicated by a thin frame around an area in the document window. To set focus to the desired area, click somewhere in it.

Browse Device Patches...

This menu item allows you to select a new Patch for a device. The menu item reflects which device is selected - in other words, you must select the device for the corresponding Browse Patches item to appear on the Edit menu.

When you select the menu item, the Browser dialog appears, allowing you to locate and select the patch, on the hard disk or within a ReFill.

When you select a patch, the device's parameters will be set according to the values stored in the patch, and the name of the patch will be shown in the patch name display. As with any change you make, this operation can be undone.

! **Any parameter adjustments you make on the device panel after selecting a patch will not affect the actual patch file (for this you need to save the patch).**

If referenced samples are missing

Patches for the Redrum, NN-19 and NN-XT contain references to samples. Just like patches, samples can be independent files on the hard disk or elements within a ReFill or a SoundFont. However, if sample files have been moved or renamed after a patch was saved, the sample file references in the patch will not be accurate.

If this is the case when you select a patch, the program will tell you so. You can then choose to either manually locate the missing files, to have the program search for them in the database and ReFills or to proceed with missing sounds.

Browse ReCycle/REX Files...

This menu item is used to add a loop to the selected Dr.Rex device. Files to be imported can be in REX, RCY or REX2 file format.

Loading a new REX file will replace any currently loaded file.

Browse Samples...

This menu item lets you load samples into the devices that use them; the Redrum, the NN-19 and the NN-XT.

The following sample formats can be loaded:

- ➔ **Wave (.wav)**
This is the standard audio format for the PC platform.
- ➔ **AIFF (.aif)**
This is the standard audio format for the Mac platform.
- ➔ **SoundFont samples (.sf2)**
This is an open standard format for wavetable synthesized audio, developed by E-mu systems and Creative Technologies.
- ➔ **REX file slices (.rex2, .rex, .rcy)**
REX files are music loops created in the ReCycle program. This program "slices" up loops into several, separate samples. These samples - or slices - can be loaded into the devices mentioned.

Redrum

To use this menu item to load a new drum sound into Redrum, proceed as follows:

1. **Select a channel in the drum machine, by clicking its Select button.**
2. **Select Browse Samples.**
The Redrum sample browser opens.
3. **Navigate to a location containing any of the sample formats listed above, select one and click Open.**

NN-19

This menu item can also be used to add a sample to a key zone in a key map in the NN19 sampler.

1. **Select a key zone.**
This can be empty, or contain a sample - it doesn't matter for now.
2. **Use the browser to add one or several (see below) sample(s).**

The following will happen:

- ➔ **If the zone contained a sample prior to loading, this will be replaced, both in the zone and in the sample memory, unless the sample was also used by another key zone.**
- ➔ **If you loaded several samples, one of the samples (the sample that was selected furthest down in the Browser list) will be loaded into the key zone, and the other samples will be loaded into the sample memory.**

NN-XT

This menu item is used for adding one or more sample(s) to a key map in the NN-XT:

1. **Make sure the Remote Editor panel is folded out, by clicking the small arrow in the bottom left corner.**
If the remote editor panel is folded in, you will only be able to browse for NN-XT patches.
 2. **Use the sample browser to add one or several sample(s).**
The sample(s) will be placed in separate zones and mapped across the same key range.
- ! **If a key map already contains a zone with a sample in it, and this is selected prior to loading, it will be replaced if you load a new, single sample. If you load several samples at once though, they will instead be added below any already loaded samples.**

Automap Samples

This menu item applies to the NN 19 Sampler. If you have a number of samples that belong together but haven't been mapped to key zones, you can use the "Automap Samples" function. This is used in the following way:

1. **Select all samples that belong together and load them in one go, using the sample browser.**
One of the samples will be loaded to a key zone spanning the whole range, and the rest will reside in the sample memory.
2. **Select Automap Samples from the Edit menu.**

Now the samples currently in memory will be arranged automatically so that:

- **Each sample will be placed correctly according to its root note, and will be tuned according to the information in the sample file.**
Most audio editing programs can save root key information as part of the file.
- **Each sample will occupy half the note range to the next sample's root note.**
The root key will always be in the middle of each zone, with the zone extending both down and up in relation to the root position. Hence, no key zone high or low limits have to be manually set!

Mapping Samples Without Root Key or Tuning Information

Some samples may not have any information about root key or tuning stored in the file (nor indicated in the file name). If this is the case, you can still make use of the Automap function:

1. **Select all samples that belong together and load them in one go, using the sample browser.**
One of the samples will be loaded to a key zone spanning the whole range, and the rest will reside in the sample memory.
2. **Manually set the root key, and adjust the tune knob if the sample needs pitch fine-tuning.**
Without any information stored in the file, or if the file name doesn't indicate the root key, you will have to use your ears for this step. Play the sample across different areas of the keyboard and listen to where it sounds the most "natural". As long as you are in the general area of the correct root key, the result should be o.k. You can always adjust this later.
3. **Select the next sample using the Sample knob, and repeat the previous step.**
Proceed like this until you have set a root key for all the samples.
4. **Select "Automap Samples" from the edit menu.**
The samples will be automatically mapped according to their set root key positions!

Delete Sample/Remove Sample

Redrum

- **To remove a sample from a Redrum drum machine, select its drum sound channel and then select "Delete Sample" from the Edit menu.**
The sample is removed from the drum sound channel and from sample memory.

NN-19

- **To remove a sample from an NN-19 Sampler's memory, select the zone it belongs to, and then select "Delete Sample" from the Edit menu.**
The sample is removed from the zone and from sample memory.

NN-XT

- **To remove a sample from an NN-XT Sampler's memory, select the zone it belongs to, and then select "Remove Samples" from the Edit menu.**
The sample is removed from the zone and from sample memory. The zone still remains though. To delete a zone, you must use the option "Delete Zones".

Delete Unused Samples

This menu item is used for the NN-19 Sampler. When you select it, all samples that are not assigned to a key zone are deleted from sampler memory.

This way you can make sure that you are not wasting any sample memory for samples that are not actually used.

Split Key Zone

This menu item is used for the NN-19 Sampler. It splits the currently selected key zone in the middle. The new zone is the upper half of the split, and is empty. The dividing point has a "handle" above it.

Delete Key Zone

This menu item is used for the NN-19 Sampler. It deletes the currently selected key zone from the key map.

Copy REX Loop to Track

This menu item is used for the Dr. Rex loop player device. To be able to make your REX loop start at the same time as other sequencer or pattern data, you "convert" the slices in the loop to notes in the sequencer:

1. **Select a sequencer track connected to the Dr.Rex device.**
2. **Set the left and right locators to encompass the section you want to fill with REX notes.**
You may want to make sure that this area doesn't contain any notes already, to avoid confusion.
3. **Select the Dr. Rex player, so that it has focus.**
4. **Pull down the Edit menu and select "Copy REX Loop to Track".**
Now, the program will create a note for each slice, positioned according to the timing of the slices. The notes will be pitched in semitone steps, with the first note on C1, the second on C#1 and so on, with one pitch for each slice. If the area between the locators is longer than the loop length, the loop notes will be repeated to fill out the loop.

Now you can reorder, overdub onto, and otherwise edit the note data, using the REX or Key edit lanes in the sequencer.

Copy Pattern to Track

This menu item is used for the Redrum drum machine and Matrix pattern sequencer. It converts the selected pattern to notes on a sequencer track. Proceed as follows:

1. **Select a sequencer track connected to the Redrum/Matrix.**
 2. **Set the left and right locators to the desired range or length.**
If the range set is longer than the pattern(s), the data will be repeated to fit the range.
 3. **Select the pattern device, so that it has the focus.**
 4. **Pull down the Edit menu and select "Copy Pattern to Track".**
Notes will be created between the left and right locators, according to the selected pattern.
- ! When copying Matrix patterns, only the Gate and Keys values will be included!**
- **If you copied a Redrum pattern, you may want to turn off the "Enable Pattern Section" before playing back the new track data.**
Otherwise, both the main sequencer and the pattern sequencer will play the drum sounds, simultaneously.

→ **If you copied a Matrix pattern, you need to connect the track to an instrument device (such as the device which was originally controlled by the Matrix), since the Matrix in itself doesn't produce any sound.**

Furthermore, you may want to disconnect the Matrix (or even remove it), to avoid having both the Matrix and the sequencer notes playing at the same time.

Shift Pattern Left/Right

These menu items are used for the Redrum and Matrix respectively.

The Shift Pattern functions move the notes in a pattern one step to the left or right.

Shift Drum Left/Right

These menu items are used for the Redrum.

The Shift Drum functions move the notes for the selected instrument one step to the left or right.

Shift Pattern Up/Down

These menu items are used for the Matrix.

The Shift Pattern functions will transpose all the notes in a pattern one semitone up or down.

! This function does not alter the Curve CV.

Randomize Pattern

This menu item is used for the Redrum and Matrix.

The Randomize Pattern function create random patterns. These can often be great starting points and help you get new ideas.

! Note that for the Matrix, Randomize affects both the Gate, Note and Curve CV!

Randomize Drum

The Randomize Drum functions creates random patterns for the selected drum sound channel in the Redrum drum machine.

Alter Pattern

This menu item is used for the Redrum and Matrix.

The Alter Pattern function modifies existing patterns. Note that there must be something in the pattern for the function to work on - using an Alter function on an empty pattern will not do anything.

! Note that for the Matrix, Alter affects both the Gate, Note and Curve CV!

Alter Drum

The Alter Pattern function modifies existing patterns for the selected drum sound. Note that there must be something in the pattern for that channel for the function to work - using an Alter function on an empty pattern will not do anything.

Auto-route Device

Auto-routing is when devices' audio and CV/gate connections are automatically routed according to default rules. Auto-routing is normally performed:

- When a new device is created.
- When moving, duplicating or pasting devices with [Shift] pressed.

However, if a device is already in the rack, you can “force” it to be auto-routed by selecting it and then select this menu item.

For more information about auto-routing rules, see [page 37](#).

Disconnect Device

This disconnects all audio and CV/gate connections from the selected device(s).

Duplicate Track

This creates a copy of the selected track, complete with all events. The duplicated track will appear below the original track in the track list.

Group

This puts a selection of events in the Arrange view into a Group:

1. Select the events that you want to Group.

It doesn't matter which lanes you select - all notes, pattern changes and controllers within the area will be included in the Group.

→ If you select events on several tracks, one Group for each track will be created.

Each Group can only contain events on one track.

2. If you want the Group to have a specific length, activate Snap and select an appropriate Snap value.

Often it is practical to create Groups that are one or several whole bars of length.

3. Select Group from the Edit menu.

Ungroup

This menu item is used to dissolve a Group:

1. Select the Group.

2. Select Ungroup from the Edit menu.

Find Identical Groups

This command helps you locate all Groups with the same contents:

1. Select a Group.

2. Select “Find Identical Groups” from the Edit menu.

All Groups with the same contents are selected in the Arrange View.

Insert Bars Between Locators

This function inserts an empty area between the locators in the main sequencer. All events after the left locator are moved to the right to “make room” for the inserted area.

Remove Bars Between Locators

This function removes all material between the locators in the main sequencer. All events after the right locator are moved to the left to “fill out” the gap after the removed section.

Convert Pattern Track to Notes

If you have recorded or drawn pattern changes on a Redrum or Matrix track, you can have the whole track converted to notes, in the following way:

1. Select the track with the pattern changes.

2. Select “Convert Pattern Track to Notes” from the Edit menu or the context menu for the track.

For each bar, the corresponding pattern is converted to notes on the track (following the same rules as for the “Copy Pattern to Track” function). The track will play back just the same as when you played the pattern device with the pattern changes (including the Pattern Enabled/Mute switch).

→ After the operation, all pattern changes are automatically removed from the track.

Redrum notes

- The “Enable Pattern Section” switch is automatically turned off when you use this function.

Matrix notes

- After performing "Convert Pattern Track to Notes", you need to move the contents to another track, or re-route the track to another device. Having the track connected to the Matrix itself is pointless, since the Matrix cannot produce any sound.
- You may want to disconnect or even remove the Matrix after performing this function.
This is because you probably don't want both the Matrix and the sequencer notes to play back at the same time.

Get User Groove

You can create your own groove and apply this using Groove Quantize:

1. **Create or record a rhythmic note "pattern" of some kind.**
You may for example record a drum pattern, or use the notes playing the slices in a REX loop.
 2. **Select the notes you want to include in the user groove.**
The groove can be of any length, but it's usually most practical to make it one or two bars long.
 3. **Select "Get User Groove" from the Edit menu or sequencer context menu.**
Your pattern is stored as the User groove.
 4. **Select any notes you want to quantize, make sure "User" is selected as Quantize value, and quantize as usual.**
The rhythmic feel of your groove is applied to the notes.
- ! The User Groove is only stored temporarily - it isn't included when you save your Song.**

Clear Automation

To remove all automation for a controller, select "Clear Automation" from the Edit menu.

This requires that the controller subtrack has focus. Click in the subtrack if you are uncertain.

Selecting "Clear Automation" will remove all controller values from the subtrack, and the text "Not Automated" will be shown.

Quantize Notes

In Reason, you use the Quantize function in the following way:

1. **Select the notes you want to quantize.**
Only notes will be affected, so you can select Groups or complete Tracks if you like.

2. **Pull down the Quantize pop-up menu on the sequencer toolbar and select a Quantize value.**

This determines to which note values the notes will be moved when you quantize. For example, if you select sixteenth notes, all notes will be moved to (or closer to) the closest sixteenth note position.

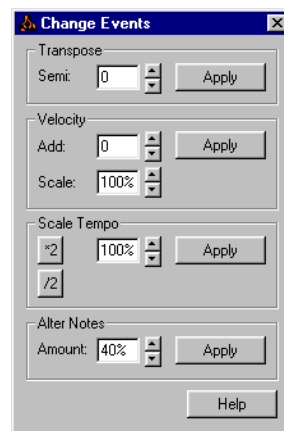
3. **Select a value from the Quantize Strength pop-up menu.**

This is a percentage, governing how much each note should be moved. If you select 100%, notes will be moved all the way to the closest Quantize value positions; if you select 50%, notes will be moved half-way, etc.

4. **Click the Quantize button or select "Quantize Notes" from the Edit menu.**

The selected notes are quantized.

Change Events...



The Change Events dialog contains some special editing functions. Proceed as follows:

1. **Select the events to which you want to apply the editing functions.**
The Change Events functions are mainly used with notes, but the Scale Tempo function will also affect controllers and pattern changes (see below).
2. **Select Change Events from the Edit menu or the context menu for the selected events.**
The Change Events dialog appears.

3. **Make settings for one of the functions in the dialog and click the Apply button next to the settings.**
All settings can be made by clicking the spin controls or by clicking in a value field and entering a value numerically. The functions are described below.
4. **If you like, use other settings in the same way.**
You can use the transport controls as usual while the dialog is open. This allows you to play back the events to check out the changes.
5. **When you are done, close the dialog.**

Transpose

This function transposes the selected notes up or down, by the specified number of semitones.

Velocity

Adjusts the velocity of the selected notes.

- **The Add field lets you add a fixed amount to the velocity values.**
To subtract, enter a negative amount. Note that the possible velocity range is 0-127. Adding an amount to a note with velocity 127 will not make any difference.
- **The Scale field allows you to scale velocities by a percentage factor.**
Scaling with a factor above 100% will increase the velocity values, but also make the difference between soft and hard notes bigger. Scaling with a factor below 100% will decrease the velocity values, but also make the difference between soft and hard notes smaller.
- **By combining the Add and Scale functions, you can adjust the “dynamics” of the notes in various ways.**
For example, by using a Scale factor below 100% and Add a suitable amount, you can “compress” the velocity values (decreasing the difference between the velocity values without lowering the average velocity).

Scale Tempo

This function will make the selected events play back faster (Scale factor above 100%) or slower (Scale factor below 100%). This is achieved by changing the position of the events (starting from the first selected event) and adjusting the length of the notes accordingly.

- **The buttons [“2] and [“/2] are “shortcuts” to Scale factors 200% and 50%, respectively.**
These are probably the most common values used, simulating double tempo and half tempo.

! This function affects all types of events: notes, controllers and pattern changes!

Alter Notes

This function alters the properties pitch, length and velocity of the selected notes in a random fashion.

- **The function will only “use” values that already exist among the selected notes.**

For example, if you have selected notes within a specific pitch interval, the altered notes will remain within this pitch interval. Similarly, only velocity values and note lengths that were already used in the selection will be applied by the Alter function. You could say that the function “shuffles” the existing properties in a selection and redistributes them among the notes.

! This means that the less variation there is among the selected notes, the less the effect of the Alter function.

- **You can adjust the amount of Altering with the Amount value.**

★ **This function is especially useful for experimenting with REX loops.**
Select some notes on a Dr.Rex track and use Alter Notes to create instant variations, without losing the timing and rhythmic feel of the loop!

Reload Samples

This menu item is used with the NN-XT sampler. When you select this, any changes you have made on a loaded sample using the single adjustment sample parameters are immediately undone, and the settings revert back to the original.



Single adjustment parameters

Add Zone

This menu item is used with the NN-XT sampler. It is used for adding an empty zone to the key map. An empty zone can be resized, moved and edited in the same way as zones that contain samples.

An empty zone is indicated with the text “***No Sample***”. After you have added an empty zone, you can assign a sample to it.

Copy Zones

This menu item is used with the NN-XT sampler. It copies the selected zone(s), and all of its settings - including references to any sample it may contain - and places it in the clipboard buffer. You can then select "Paste Zones" to create a new zone that is an exact replica of the copied zone(s). Note that copying/pasting zones can also be performed between NN-XT devices.

Paste Zones

This menu item is used with the NN-XT sampler. If you have used the "Copy Zones" command, with any number of selected zones, you can create exact duplicates of these by using the "Paste Zones" command. The pasted zones will then be added below any existing zones in the key map.

Duplicate Zones

This menu item is used with the NN-XT sampler. It lets you duplicate any number of already existing zones (containing samples or empty).

1. Select the zone(s) you want to copy.

2. Select "Duplicate Zones".

The selected zones will now be copied and automatically inserted below the last one in the key map display.

The duplicated zones will contain references to the same samples as the original zones. They will also have the exact same key ranges and parameter settings.

Delete Zones

This menu item is used with the NN-XT sampler. Selecting this option will remove both the selected zones, and any samples they may contain.

Select All Zones

This menu item is used with the NN-XT sampler. This option will automatically select all zones in a key map.

Copy Parameters to Selected Zones

This menu item is used with the NN-XT sampler. It lets you easily copy parameter settings from one zone to any number of other zones. Proceed as follows:

1. Select all the zones you want to involve in the operation.

By this we mean the zone with the settings you wish to copy, and the zone(s) to which you want to copy the settings.

2. Make sure the zone that contains the settings you want to copy has edit focus by clicking on it.

3. Select "Copy Parameters to Selected Zones".

All the selected zones will now get the exact same parameter settings.

! Observe that this only applies to the synth parameters (LFOs, envelopes etc.). Sample parameters (root key, velocity range etc.) can not be copied.

Sort Zones by Note

This menu item is used with the NN-XT sampler. This option lets you automatically sort the selected zones *within a Group* in descending order according to their key ranges.

When you invoke this option, the selected zones will be sorted from top to bottom in the display starting with the one with the lowest range.

If two or more zones have the same key range, they are instead sorted by velocity range.

Sort Zones by Velocity

This menu item is used with the NN-XT sampler. This option lets you automatically sort the selected zones *within a Group* in descending order according to their set low or high velocity values.

When you invoke this option, the selected zones will be sorted from top to bottom starting with the one with the highest "Lo Vel" value.

If two or more zones have the same velocity range, they are instead sorted by key range.

Group Selected Zones

This menu item is used with the NN-XT sampler. It lets you put any number of selected zones together in a group.

Grouping zones is good for two things:

→ To allow you to quickly select a number of zones that "belong together."

For example if you have created a layered sound consisting of piano and strings, you could put all string samples in one group and all piano samples in one group. Then you can quickly select all piano samples and make an adjustment to them by trimming a parameter.

→ To group zones that need to share group settings together.

For example, you may want to set a group to legato and monophonic mode and add some portamento so that you can play a part where you slide between notes.

Proceed as follows:

1. Select the zones you want to group together.

The zones don't have to be contiguous in order to be grouped. Regardless of their original positions in the samples column, they will all be put together in succession.

2. Select "Group Selected Zones".

The zones are grouped.

Note that there is always at least one group, since the zones you create are always grouped together by default.

Set Root Notes from Pitch Detection

This menu item is used with the NN-XT sampler. All instrument sounds have an inherent pitch. When playing a sample of such a sound on the keyboard, the keys you play must correspond to that pitch. For example, you may have recorded a piano playing the key "C3". When you map this onto the NN-XT key map, you must set things up so that the sampler plays back the sample at original pitch when you press the key C3, and this is done by adjusting the root note.

The NN-XT features a pitch detection function to help you set the root keys of loaded samples. This is useful if you for example load a sample that you haven't recorded yourself, and you don't have any information about its original pitch.

Proceed as follows:

1. Select all the zones you want to be subject to pitch detection.

2. Select "Set Root Notes from Pitch Detection".

The samples in all the selected zones will now be analyzed, and the detected root keys will automatically be set for you.

! Note that for this to work properly, the samples must have some form of perceivable pitch. If it is sampled speech, or a snare drum for example, it probably doesn't have any discernible pitch.

Automap Zones

This menu item is used with the NN-XT sampler. The automap function can be used as a quick way of creating a key map, or as a good starting point for further adjustments of a key map.

Automap works under the assumption that you intend to create a key map for a complete instrument, for example a number of samples of a piano, all at different pitches.

1. Load the samples you want to Automap.

Now you have three options:

- Trust that the root note information in the files is already correct.
- Manually adjust the root notes (and tuning) for all the samples.
- Use "Set Root Notes from Pitch Detection" to automatically set up the root notes.

2. Select all zones you want to automap.

3. Select Automap Zones.

All the selected zones will now be arranged automatically in the following way:

- **The zones will be sorted in the display (from top to bottom - lowest key first) according to the root keys.**
- **The zones will be assigned key ranges according to the root keys.**
The key ranges are set up so that the split between two zones is exactly in the middle between the zones' root notes. If two zones have the same root key they will be assigned the same key range.

Create Velocity Crossfades

This menu item is used with the NN-XT sampler. This is used for automatically setting up velocity crossfades for smooth transitions between overlapping zones. To set up crossfades, you adjust the fade out and fade in values for the overlapping zones.

An example:

→ Two zones are both set to play in the full velocity range of 1-127.

→ Zone 1 has a fade out value of 40.

This means that this zone will play at full level with velocity values below 40. With higher velocity values, it will gradually fade out.

→ Zone 2 has a fade in value of 80.

This has the effect that as you play velocity values up to 80, this zone will gradually fade in. With velocity values above 80, it will play at full level.

Instead of manually setting up a crossfade, you can let NN-XT do it for you. Proceed as follows:

1. Set up the zones so that their velocity ranges overlap, as desired.

2. Select the zones.

You can select as many zones as you wish, not just one pair of overlapping zones.

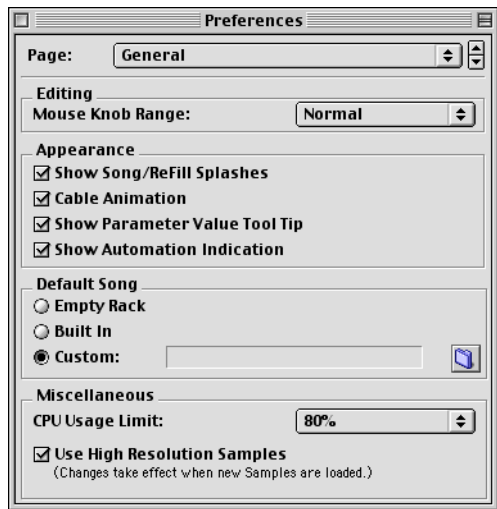
3. Select "Create Velocity Crossfades".

NN-XT will analyze the overlapping zones and automatically set up what it deems to be appropriate fade in and fade out values for the zones.

Note the following important points:

- **This operation will not work if both zones have full velocity ranges.**
At least one of the zones must have a partial velocity range (see page 170).
- **This operation will not work if the zones are completely overlapping.**

Preferences – General



Mouse Knob Range

This lets you adjust the response sensitivity of the various knobs in Reason when manipulating them with the mouse. A higher sensitivity gives a higher degree of precision. You can choose between Normal, Precise and Very Precise.

Show Song/ReFill Splashes

This option allows you to decide whether the Browser should display the Splash Pictures or not when Browsing for a song or a ReFill. Furthermore, if this option is deactivated, song splashes will not be shown when you open songs.

Cable Animation

Cables in Reason are animated in a lifelike fashion when flipping the Rack and making connections. Should you so wish, you can choose to disable the cable movement animation by deactivating this checkbox.

Show Parameter Value Tool Tip

Normally, if you hold the mouse pointer over a parameter on a device panel for a moment, a Tool Tip appears displaying the name and the current value of the parameter. If you uncheck this option, Tool Tips will not be displayed.

Show Automation Indication

If a parameter is automated in the sequencer, this is, by default, indicated by a colored square around the parameter on the device panel. If you uncheck this option, automation will not be indicated.

Default Song

Every time you start Reason, and every time you select “New” from the File menu, a default song opens. The “standard” default song contains a few select devices.

This section lets you decide exactly what you want the default song to look like, by using the radio buttons to the left:

- **Empty Rack** - This is an empty rack. Well, almost empty, since it contains the Reason hardware interface.
- **Built In** - This is a built-in Reason Song, containing a few devices. Note that it is not possible to open this song by regular means - via the browser - since it is not an “independent”.rns-file, and thus does not reside anywhere in the Reason folder.
- **Custom** - This allows you to select a custom default song. Any Reason song can be used, so if you often create songs using the same or similar device setups, you can use a previously created song as the default song. This way, all new songs you create will have the same device setup.

To customize the contents of new songs, proceed as follows:

1. **Select New from the File menu to create a new song document window.**
2. **Add/remove devices and make settings as desired.**
Typically, you may want the default song to contain your choice of devices and possibly some patterns. You could also make some special routing between devices, or even add some sequencer data.
3. **Save the song anywhere you like (preferably in the Reason program folder though) and under any name.**
4. **Pull down the Edit menu and open the Preferences dialog.**
5. **Go to the General page, and under “Default Song” click the radio button marked “Custom”.**
6. **Click the browser button to the right in the dialog, navigate to the song you saved earlier and click “Open”.**
The name of the song appears in the textbox in the dialog.

7. Close the Preferences dialog.

The next time you launch the program or select New from the File menu, the new song document will contain the devices and settings you made.

CPU Usage Limit

Reason is a powerful program but also demanding in terms of processing power. The more devices you add to your rack, the more of your computer's resources it will use.

Furthermore, as you use more and more of your computer resources for creating audio, less will be available for the user interface, resulting in slower performance in terms of graphics and overall responsiveness.

The CPU Usage Limit setting allows you to set a limit on how much of the CPU (computer processor) that can be used for creating audio. The remaining capacity is reserved for the user interface and the graphics.

Set this so that you feel comfortable using the program, even when a very demanding song document is played back.

Use High Resolution Samples

Reason has the capability to play back samples with practically any resolution. This means that if for instance 24-bit samples are loaded in a sampler or the ReDrum, playback of the samples can be in 24-bit resolution as well. If you are using such samples and want Reason to play them back in their original high resolution, make sure that this checkbox is ticked.

If this is activated, and if your audio card supports it, Reason will play back high resolution samples in their original resolution. If this option is not activated, Reason will play back all samples in 16-bit resolution, regardless of their original resolution.

Preferences – Audio



Master Tune

This lets you adjust the global tuning in Reason. Standard tuning is "middle A" at 440 Hz. You can adjust this by +/- 100 cents.

Audio Card Driver – Windows

This menu lists all the available Audio Card Drivers on your system, and lets you select which one Reason should use. Which option to select depends on the audio hardware:

→ If you are using audio hardware for which there is a specific ASIO driver, you should select this.

With an ASIO driver written specifically for the audio hardware you will get lower latency (see below), support for higher sampling frequencies (up to 96 kHz in 24 bit/32 bit float resolution), and possibly better support for additional hardware features such as multiple outputs.

→ **If there is no specific ASIO driver, you should select the Direct Sound driver for the audio hardware.**

This makes Reason communicate with the hardware via Direct Sound (a part of the Microsoft DirectX package). For this to be possible, you need to have DirectX installed on your computer, and there must be a Direct Sound driver for the audio hardware.

→ **If the audio hardware doesn't support Direct Sound (i.e. there is no Direct Sound driver for the audio hardware), select the MME driver for the audio hardware.**

This makes use of Windows Multimedia Extensions, the part of Windows that handles audio, MIDI, etc. Using MME often results in larger latency values (see below).

Audio Card Driver – Mac OS X

This menu lists all the available Audio Card Drivers on your system, and lets you select which one Reason should use. Which option to select depends on the audio hardware:

→ **Normally, you should select one of the driver options that start with the word “CoreAudio”.**

Select the option that corresponds to the hardware you want to use (the built-in audio connectors or some additional audio hardware that you have installed).

→ **Other options may be available, mainly for compatibility with all possible hardware/software configurations.**

You should use these only when required.

Audio Card Driver – Mac OS 9

This menu lists all the available Audio Card Drivers on your system, and lets you select which one Reason should use. Which option to select depends on the audio hardware:

→ **If you are using audio hardware for which there is a specific ASIO driver, you should select this.**

With an ASIO driver written specifically for the audio hardware you will get lower latency (see below), support for higher sampling frequencies (up to 96 kHz in 24 bit/32 bit float resolution), and possibly better support for additional hardware features such as multiple outputs.

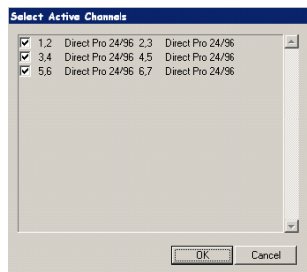
If there is no specific ASIO driver for your audio hardware, you will use the Apple Sound Manager. This is the sound driver protocol that comes with the Mac OS, and Reason communicates with the audio hardware using this.

→ **If you plan to use the internal audio outputs on your computer, please select “SM Built-in”.**

→ **If you have some additional audio hardware (such as USB-speakers) installed, please select “SM Device Name”, where “Device Name” is the name of your audio hardware.**

Active Channels (ASIO and CoreAudio Only)

This displays the number of audio channels (outputs) the currently selected audio hardware supports. For a regular stereo card, this number will be “2”. If your audio card has multiple outputs and an ASIO or CoreAudio driver is selected for it, the “Channels” button will be available. By clicking on it, you will be able to select which channel outputs (stereo pairs) should be active. Active outputs will be indicated in the Reason Hardware Interface.



Clock Source (ASIO Only)

If you are using an ASIO driver for your audio hardware, you have the possibility of selecting a Clock Source. This is used for determining the source to which audio playback should synchronize its sample rate. If you have an audio card and a driver that supports it, you can choose to synchronize to external sources.

ASIO Control Panel (ASIO Only)

If you have selected an ASIO driver, this button brings up a control panel window specifically for that audio hardware. This may contain buffer settings, routing options, synchronization alternatives etc.

Sample Rate

This lets you specify the playback sample rate. The options available on this menu depends on which sample rates are supported by your audio hardware.

Play in Background

When this is activated, Reason will not “release its grip” on the audio hardware when another application is active.

- The advantage is that Reason will keep playing while you work in the other application.
- The disadvantage is that other audio applications may not be able to play any audio, depending on the type of driver used.

Output Latency & Buffer Size

The Output latency is the delay between when audio is “sent” from the program and when you actually hear it. The latency in an audio system depends on the audio hardware, its drivers and their settings.

If the latency is large, you will notice that the sound is delayed when you play a device from a MIDI keyboard. You may also notice that reactions are delayed when adjusting controls on the device panels (for example, if you lower the volume of a device, you will not hear this immediately but after the latency time). Therefore, you want to get as low a latency value as possible.

When you select a driver, its latency value is automatically reported and displayed in the Preferences-Audio dialog. Depending on the audio hardware and the driver, you may be able to adjust this value:

→ **If you are running Reason under Windows using a Direct Sound or MME driver, or Mac OS X using a CoreAudio driver, you can adjust the latency value by using the Buffer Size slider or the up/down arrow buttons. The highest and lowest possible values depend on the driver.**

- **If you are using an ASIO driver specifically written for the audio hardware, you can in most cases make settings for the hardware by clicking the Control Panel button. This opens the hardware's ASIO Device Control Panel, which may or may not contain parameters for adjusting the latency. Usually this is done by changing the number and/or size of the audio buffers - the smaller the audio buffers, the lower the latency. Please consult the documentation of your audio hardware and its ASIO drivers for details!**
- **If you are running Reason on a Mac using the Sound Manager driver protocol, you cannot change the latency.**

OK, so why not just set the latency to the lowest possible value? The problem is that selecting too low a latency is likely to result in playback problems (clicks, pops, dropouts, etc.). There are several technical reasons for this, the main one being that with smaller buffers (lower latency), the average strain on the CPU will be higher. This also means that the more CPU-intensive your Reason song (i.e. the more devices you use), the higher the minimum latency required for avoiding playback difficulties.

Latency Compensation

This control should normally only be adjusted when synchronizing Reason to external MIDI Clock.

Because of the latency problem, you might need to adjust Reason's playback in relation to the MIDI Clock sync master, so that they are in perfect time. The tempo will not differ between the two, but Reason might play ahead or behind the other application. You might need to adjust this. However, this is something you only need to do once. The setting is stored with your other preferences, so you don't need to adjust it again.

Proceed as follows:

1. **Set up the other application so that it generates a solid click, on for example quarter or eighth notes, preferably with a special sound on the downbeat. This click can either come from an internal metronome or from a MIDI source. If you use a MIDI source, make sure you pick one that has solid MIDI timing.**
2. **Set up Reason so that it plays a similar rhythm as the other application. You might for example use the Redrum drum computer for this.**
3. **Start the two applications in sync.**
4. **Make sure you hear both applications at approximately equal level.**
5. **Open the Preferences dialog in Reason and select the Audio page.**
6. **Trim the "Latency compensation" setting until the "clicks" from the both sources sound at exactly the same time.**
7. **Close the Preferences dialog in Reason.**

Preferences – MIDI

Sequencer Input & Channel

The Sequencer is the "standard" port for receiving MIDI input. This is what you should be using if you intend to use the Reason sequencer.

Once you have selected your MIDI interface on the Sequencer Port pop-up (and which channel it should receive on), you can direct incoming MIDI to any device by just clicking the "In" column to the left of a track name in the track list.

Preferences – Advanced MIDI

External Control Bus Inputs

The External Bus inputs provide up to 64 MIDI input channels divided into four buses, each with 16 channels.

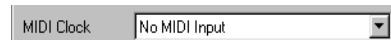
- ➔ **These MIDI inputs are primarily for controlling Reason Devices from an external sequencer.**

This could be an external hardware sequencer or sequencer software that is installed on the same computer as Reason. You should preferably use a multiple port MIDI interface, so you can select separate ports for Reason and the other MIDI devices to use, although this isn't strictly required. See the chapter "Routing MIDI to Reason" in the electronic documentation.

Remote Control Input

The Remote Control input is used for assigning a MIDI port for receiving MIDI Controller messages. How to use Remote Control is described in the electronic documentation in the chapter "MIDI and Keyboard Remote Control".

MIDI Clock Input



Using MIDI Clock, you can slave (synchronize) Reason to hardware devices (tape recorders, drum machines, stand alone sequencers, workstations etc.) and other computer programs running on the same or another computer. MIDI Clock is a very fast "metronome" that can be transmitted in a MIDI Cable. As part of the MIDI Clock concept there are also instructions for Start, Stop and locating to sixteenth note positions.

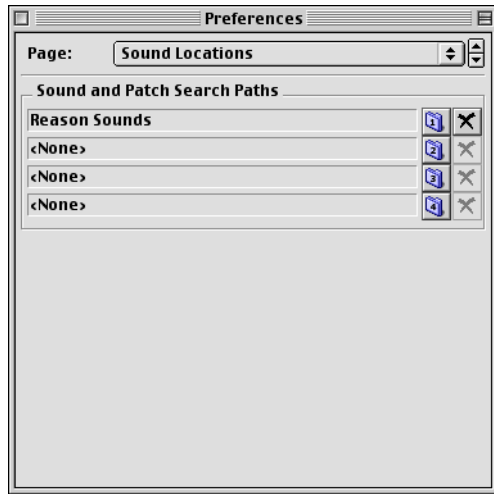
- ➔ **By first selecting the appropriate MIDI input using the MIDI Clock pop-up and then selecting "MIDI Clock Sync" on the Options menu, Reason is made ready to receive MIDI Clock sync.**
See the "Synchronization" chapter for more information.

Disable MIDI Priority Boost (Windows & Mac OS 9 only)

Reason normally tries to trim your computer system so that MIDI Input gets a higher priority than it normally does. This is to ensure best possible performance when for example recording notes via MIDI.

However, we can not guarantee that this attempt to boost MIDI priority will work on all systems and with all MIDI Interfaces. If you run into problems with your MIDI, try activating this switch.

Preferences – Sound Locations



4. If you like, specify search path 2 to 4 in the same way.

It is normally enough to specify a single path, since all underlying folders are automatically included in the database. Use the additional paths if you use more than one hard drive, CD-ROM drives etc.

When you add sound files or save Reason files, you should place them within the database (under one of the specified search path folders).

Sound and Patch Search Paths

Reason songs and patches can contain references to other files on your hard disk, such as samples. To keep track of all files, Reason makes use of a "database". If you keep your Reason files within the database, Reason can update file paths, automatically search for missing files, etc.

This database consists of up to four different folders on disk (and all their sub-folders). You specify which folders to use as database in the following way:

1. Click the "1" folder button below the heading "Sound and Patch Search Paths".

A file dialog appears.

2. Navigate to the desired folder and select it.

You can select a folder on any drive (including mapped network drives under Windows).

3. Click OK.

The folder is added as the first search path in the database.

Create Menu

Sequencer Track

Tracks are automatically created when you create instrument devices in the rack. Still, you may need to create additional tracks (e.g. for recording effect device automation):

- **To create a new sequencer track, pull down the Create menu and select Sequencer Track.**
The new track will appear below the currently selected track in the track list. Initially, it will not be connected to any device.
- **You can also create a new sequencer track specifically for a device by using the Create Sequencer Track for Device item on the device's context menu.**
This works the same as when creating a new device, i.e. the new track is connected to the device and has the same name.

Device List

To create a new device, select the desired item on the Create menu.

- **The new device is added directly below the currently selected device in the rack.**
If no device is selected, the new device is added at the bottom of the rack.
- **When you add a new device, Reason attempts to route it in a logical way.**
- **A new track will automatically be created in the sequencer, and routed to the new device.**
The track will have the same name as the device. MIDI input will also automatically be set to the new track, allowing you to immediately play the created device via MIDI.
- ! **By default, this only applies to instrument devices, not to mixers or effect devices. If you hold down [Option] (Mac) or [Alt] (Windows) when you create the device, the opposite is true, i.e. mixers and effect devices get new tracks but instrument devices don't.**

Options Menu

Internal Sync/MIDI Clock Sync/ReWire Sync

These three options are used to specify which type of tempo synchronization you prefer:

Internal Sync

When this is activated, the program is not synchronized to any external source. It plays in the tempo set on the transport panel.

MIDI Clock Sync

When this is activated, the program is synchronized to external MIDI Clock, as set up in the Preferences dialog. The tempo setting on the Transport is of no relevance, Reason plays in the tempo of the incoming MIDI Clock signals.

ReWire Sync

When this is activated, Reason is synchronized to another application via ReWire. This is not a setting that you can activate yourself, it is automatically enabled when the program is in ReWire slave mode.

Enable Keyboard Remote

When this is activated, keyboard keys can be used to control devices, as set up with the Edit Keyboard Remote feature.

Edit Keyboard Remote

- **To get an overview of which parameters are remote controllable select “Edit Keyboard Remote” from the Options menu.**

When done, each device you select will show a yellow arrow symbol beside every parameter that can be assigned a keyboard remote.

- **If you click on an assignable parameter, a dialog appears allowing you to select a key command for that parameter.**

You may use any key or a combination of [Shift] + any key.

- **Simply press the key (or key combination) you wish to use to remote control the parameter.**

The “Key Received” field momentarily indicates that it is “learning” the key-stroke(s), and then the dialog displays the name of the key you have pressed. If [Shift] was used, the box beside the word Shift in the dialog is ticked.

- ! **Note that the transport panel uses the numeric keypad for various commands. If you assign a parameter to a single numeric key, the corresponding transport functionality will be overridden!**

About the two Edit Keyboard Remote Modes

If Edit Keyboard Remote is enabled (ticked) on the Options menu, assigned parameters are “tagged”, showing the remote key for that parameter. In this mode, however, you cannot operate Reason normally, as every parameter you click on will open the Key Remote dialog. This mode is primarily for overview of available parameters and the current assignments.

- **Another way to assign keyboard remote commands is to have “Edit Keyboard Remote” *deselected* on the Options menu, and to simply [Ctrl]-click (Mac) / right-click (Windows) the parameter you wish to remote control.**

This opens a pop-up menu, where one of the options will be “Edit Keyboard Remote”. Selecting this opens the Key Remote dialog. Thus, you do not have to enable/disable Edit mode from the Options menu if you know that a parameter is assignable.

- ! **If you try to assign a Remote Key that is already in use, you will get an alert asking if you wish to change the current assignment.**

Clear All Keyboard Remote

This menu command removes all keyboard mapping you have set up for the song.

Enable MIDI Remote Mapping

When this is activated, MIDI messages can be used to control devices, as set up with the Edit MIDI Remote feature.

Edit MIDI Remote Mapping

1. **To get an overview of which parameters are MIDI remote controllable select “Edit MIDI Remote Mapping” from the Options menu.**

When done, each device you select will show a green arrow symbol beside every parameter that can be assigned a MIDI remote.

2. **If you click on an assignable parameter, a dialog appears allowing you to select a MIDI controller (or a Note number) to control that parameter.**

Note numbers function exactly like Keyboard remote - they can only control on/off or min/max values.

3. **Make sure that the “Learn from MIDI Input” box is ticked.**

4. **Simply turn the knob (or slider etc.) that you wish to use to remote control the parameter.**

The “MIDI Received” field momentarily flickers as you turn the knob, and then the dialog shows the controller number and the channel it is transmitted on.

5. **Click “OK” to exit the dialog.**

The selected parameter now has a tag, displaying the controller number, and the MIDI channel used.

6. **To exit Edit MIDI Remote Mapping mode, deselect it from the Options menu.**

You do not always have to use this method - see below.

About the two Edit MIDI Remote Mapping Modes

If Edit MIDI Remote Mapping is enabled (ticked) on the Options menu, assigned parameters are “tagged”, and the arrow indicators show the assignable parameters. In this mode, however, you cannot operate Reason normally, as every parameter you click on will open the MIDI Remote dialog. The Edit mode is primarily for overview of available parameters and the current assignments.

- **Another way to assign keyboard remote commands is to have “Edit MIDI Remote Mapping” *deselected* on the Options menu, and to simply [Ctrl]-click (Mac) / right-click (Windows) the parameter you wish to remote control.**

This opens a pop-up menu, where one of the options will be “Edit MIDI Remote Mapping”. Selecting this opens the MIDI Remote dialog. Thus, you do not have to select Edit mode from the Options menu if you already know that a parameter is free and assignable.

Clear All MIDI Remote Mapping

This menu command removes all MIDI Remote mapping that you have set up for the song.

Toggle Rack Front/Rear

This switches the rack between the front and rear views. A quicker way to do this is to press [Tab].

Show Cables

If you have made many connections in Reason, the cables can sometimes obscure the view, making it difficult to read the text printed on the back panels of the devices. You can hide/show all cables in the following way:

- **Select “Show/Hide cables” on the Options menu to hide all cables.**
When cables are hidden, connections are indicated by a colored connector. Repeating the above procedure makes the cables appear again.
- **When hidden, you can still connect or disconnect cables in the same way as when they are shown.**

Checking Connections

It is possible to check to which device a jack is connected, which is useful if the cables are hidden, or if the connected devices are located far apart in the rack:

- **Positioning the pointer over a connector makes a tool tip appear after a moment, showing the device and the specific connector at the other end.**

Follow Song

When this is activated, the sequencer Arrange and Edit views will scroll with the song pointer, on playback. When this item is deactivated, the view will not scroll automatically.

Windows Menu (Windows Version)

Stay on top

When this is activated, the Reason window will always stay on top of other program's windows.

Adjust frame to clients

This changes the size of the application window so that it exactly fits the document windows.

Detach/Attach Sequencer Window

Selecting this will detach the sequencer pane from the rack, and open it in a separate window. When the sequencer is detached, the menu item text changes from Detach to Attach. Selecting this will then reattach the sequencer to the rack.

Cascade

This moves and resizes the open song documents, so that they are arranged in an overlapping pattern.

Tile Horizontally

This moves and resizes the open song documents, in a horizontal pattern.

Tile Vertically

This moves and resizes the open song documents, in a vertical pattern.

Arrange Icons

If you have minimized windows and moved them around in the application window, this command cleans up their positions on screen.

Window List

This lists all open song documents. Selecting one makes it the active window.

Windows Menu (Mac OS Version)

Detach/Attach Sequencer Window

Selecting this will detach the sequencer pane from the rack, and open it in a separate window. When the sequencer is detached, the menu item text changes from Detach to Attach. Selecting this will then reattach the sequencer to the rack.

Minimize (Mac OS X only)

This minimizes a selected song document.

Window List

This lists all open song documents. Selecting one makes it the active window.

Help/Contacts Menu

Contents (Windows only)

This menu item opens up the Help system with the Contents tab selected.

Index (Windows only)

This menu item opens up the Help system with the Index tab selected.

Search (Windows only)

This menu item opens up the Help system with the Search tab selected.

Internet Page Menu Options

About the Internet menu alternatives

Regardless of which of the Internet options you select, you will be connected to the Internet using your preferred browser. The browser will then take you to the page specified in the dialog.

Go to the Propellerhead Homepage

This takes you to the main entrance on the Propellerheads web site.

Download Reason Songs

This takes you to our archives of song files that you can download and use. You can also contribute with your own creations!

Download Reason ReFills

This takes you to ReFill archives with the latest offerings in free sounds for Reason!

Reason Tech Info and Support

If you have a problem or a technical question, this is the place to go!

Order Reason Now

This allows you to purchase your own personal copy of this exquisite application!

Register Reason Now

This takes you to the Propellerhead Software registration pages. Once registered you can download free sounds, chat with other Reason users and upload songs for others to hear!

About Reason (Windows only)

This menu item opens up a dialog that informs you about the version of the program and the people behind it.



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→ About Audio on Computers

General Information

Audio Quality

The general audio quality in a computer based synthesizer system depends on two things:

→ The quality of the software calculating the audio.

In our case, this is the Reason DSP (Digital Signal Processing) code.

- Reason uses 32-bit floating point arithmetic for all internal audio operations which ensures the highest possible resolution throughout the signal chain.
- The program supports 16, 20, and 24 bit audio output.
- The program supports sampling frequencies from 22kHz to 96kHz.
- A number of digital audio techniques are implemented that reduce the risk of “aliasing”, background noise, unwanted distortion and “zipper noise”.

There is no technical reason why this program should not sound as good as or better than dedicated, professional hardware.

→ The quality of the hardware playing back the sound.

In a PC this is the sound card. In the Mac it is the built in audio circuitry or any audio card you have installed. Don't be fooled by the “16 bit, 44.1kHz, CD quality” tags. How good some audio hardware actually sounds depends on a number of things, its frequency range and frequency response curve, the signal to noise ratio, the distortion under various circumstances, etc. Furthermore, some designs are more prone to disturbance from the other electronics in the computer than others. Such disturbance might add hum or high pitched noise to the signal.

As you probably understand by now, this is a big subject and there's no way we can help you find the right solution in this manual. There are a number of text books and magazines covering this subject and any music dealer specializing in computers will happily help you out. The only advice we can give is that if you are serious about sound, choose your audio hardware carefully!

About Latency

On any personal computer system, there is a delay between the moment you “ask” the hardware to play a sound and when it actually does it. This delay is referred to as the “latency” of the design. This imposes a problem for any system where you want real time user input to affect the sound.

! See the Optimizing chapter for basic information on adjusting Output Latency!

Why is there latency?

Any audio application creates its audio in chunks. These chunks are then passed on to the audio card where they are temporarily stored before being converted into regular audio signals.

The storage place for these chunks are called “buffers” (an analogy would be a bucket brigade, where a number of people each have a bucket, and water is poured from one bucket to another to reach its final destination).

The smaller the buffers and the fewer they are, the more responsive the system will be (lower latency) However, this will also raise the demands on the computer and its software. If the system can't cope up with moving the data to and from the buffers fast enough, there will be problems that manifest themselves as glitches in audio playback.

To make things worse, audio playback always competes with other activities on your computer. For example, under Windows, an Output Latency setting that works perfect under normal circumstances might be far too low when you try to open files during playback, switch over to another program while Reason is playing or simply play back a very demanding song.

What is acceptable?

Normally, hardware synthesizers provide you with a latency of 3 to 7 ms (milliseconds – thousands of a second), at least if the instrument is targeted towards a “professional” audience.

On a regular PC or Mac, the latency can vary from anything from 2ms to 750ms! This wide range of values is an effect of the fact that computers and their operating systems were created for many purposes, not just playing back audio. For multimedia and games, a latency of a 100ms or more is perfectly acceptable, but for playing a musical instrument it is not!

- The internal audio under Mac OS 9.x provides an output latency of 11ms. This is deemed acceptable by most users.
- A regular PC “SoundBlaster” type audio card with an MME driver (see later in this chapter) might at best give you a latency of around 160ms.
- The same card with a DirectX driver provides at best around 40ms.
- A card specifically designed for low latency, with an ASIO driver, can give you figures as low as 3 ms under both Mac OS and Windows. This is just as good as any dedicated hardware synthesizer!
- The best possible situation is achieved using a fairly powerful Macintosh computer, running Mac OS X and using a CoreAudio driver. Such a setup can give as low a latency figure as 1 ms! This is better than most dedicated hardware synthesizers.



ABOUT AUDIO ON COMPUTERS

Reason's built in sequencer is not affected by latency!

When Reason's sequencer is playing back a song, the timing between notes is perfect! Once playback of a Reason pattern or song is up and running, latency isn't a consideration at all. The computer clocks the audio between the steps and does this with perfect quartz accuracy! The timing is immaculate!

ReWire and Latency

When you run Reason as a ReWire slave, it is the other program, the Rewire master that is responsible for actually creating the audio and playing it back via the audio card. This means that it is the master program's latency you will get as a final result.

! When Reason runs as a ReWire slave, what audio card you have, what driver you use, and settings you have made in the Preferences dialog are of no importance at all! All audio card settings are then instead done in the ReWire master application!

For information on ReWire, see ["Using Reason as a ReWire Slave"](#).

Reducing latency

⚙️ **Please note that internal audio under Mac OS 9 has a fixed latency of 11ms which is very stable under all normal circumstances. The tips below are for Windows users and for Mac OS 9 users with additional audio cards.**

There are a few general methods for making sure latency is as low as possible:

- ➔ **Use a card with an ASIO driver.**
While this in itself is no guarantee for low latency, ASIO drivers generally perform better than MME or DirectX.
- ➔ **Select an audio card that supports low latency (small buffers) and which is known for well written ASIO drivers.**
- ➔ **Remove background tasks on your computer.**
This might be any background utility you have installed as well as networking, background internet activities etc.
- ➔ **Optimize your songs.**
You might run into situations where you have to raise the Output Latency setting to be able to play back a very demanding song on your computer. Another option would be to actually optimize the song. See ["Optimizing Performance"](#) for details.
- ➔ **Get a faster computer.**
This is related to the point above and only required if you find that you need to increase Output Latency because your computer can't really cope with the songs you try to play.

PC Specific Information

About ASIO DirectX, MME and the Sound Buffer setting

There are three ways for Windows to access an audio card:

Via an MME (MultiMedia Extensions) driver

This system has been around since Windows 3.0, and it is this type of driver that is normally installed in the Control Panel and via Plug'n'Play. Most regular sound playback (like when Windows goes "bing" on startup) happens via MME.

- Practically all cards come with an MME driver. If your card appears in the System part of the Control Panel, you have an MME driver installed.
- Using a card via an MME driver gives you the worst latency figures, especially under Windows 98.
- Only one program at a time can use a card accessed via MME.

Via a DirectX driver

DirectX is a later system developed by Microsoft to provide developers with more efficient routines to access audio.

- Not all cards come with DirectX drivers. However, drivers for some cards are included with DirectX itself.
- Using a card via a DirectX driver gives you a shorter latency, between 40 and 90 milliseconds.
- If you use DirectX 3 or later, all programs that access the card via DirectX and make use of the DirectX "secondary buffer" feature can use it at the same time, and Reason can play in the background.

! Only use DirectX if you are sure that there is a "certified" DirectX driver installed for your sound card.

- If in doubt, contact your audio card vendor to check whether there's a DirectX driver for your card or not.

⚙️ **More information about DirectX can be found on Microsoft's DirectX web pages, at www.microsoft.com/directx.**

Via an ASIO Driver

This is your best option if it is available. More and more audio cards designed for serious music and audio use come with ASIO drivers.

As stated above, ASIO does not guarantee low latency, but it allows for it if the audio card designers take advantage of its possibilities.

- Not all cards come with ASIO drivers. If in doubt check with the audio card manufacturer.
- Using a card via an ASIO driver can give latency figures as low as 3ms.
- When you use ASIO, only one program at a time can access the card.

★ **More information about ASIO can be found on Steinberg Media Technologies' web pages, www.steinberg.net.**

Intel vs. Other Processors

When you run Reason under Windows, the clock speed of the processor is a major factor determining how many devices you can use at the same time. However, there are other factors that should be taken into account, and one important such factor is "floating point arithmetic performance".

All audio operations in Reason are done with floating point calculations (counting with decimal numbers rather than with non-decimal numbers, integers) to ensure the highest possible audio quality. You can get high audio quality on an integer system too, but floating point is effective and accurate when it is available.

Intel Pentium processors are fast at floating point mathematics. Some other lower priced processor have taken shortcuts which reduce their performance in this particular aspect. This will have noticeable effect on the performance of the program. Our advice is:

★ **If you plan to buy a computer specifically for Reason, you can play it safe and choose an Intel processor. Alternatively, make sure you select a processor that is renowned for high floating point arithmetic performance!**

Macintosh Specific Information

Mac OS X

Under Mac OS X, all communication with most audio hardware can be handled by the internal CoreAudio framework.

- **You should normally use one of the driver options that start with the word "CoreAudio".**
Select the option that corresponds to the hardware you want to use (the built-in audio connectors or some additional audio hardware that you have installed).
- **Other options may be available as well, mainly for compatibility with all possible hardware/software configurations.**
You should only use these when required.

Mac OS 9

Under Mac OS 9 there are two ways you can play back audio:

- Using the Sound Manager
- Using ASIO

What the Sound Manager does/is

The Sound Manager is a set of software routines in the Mac OS. These routines take care of everything related to sound. If you are using the internal audio on your Macintosh computer you are using the Sound Manager, it is built into the system.

One specific character of the Sound Manager is its ability to mix audio from several applications. This means that even when you run Reason, you can run other Sound Manager compatible applications at the same time, and they will all sound.

Mac Audio Cards that play back via the Sound Manager

There are a rare few Mac audio cards that play back via the Sound Manager.

- ! **If you have an audio card for your Macintosh, we strongly recommend you to try and find an ASIO driver for it instead of using a Sound Manager driver. This will give you higher reliability and performance.**

Mac Audio Cards with an ASIO Driver

An audio card with an ASIO driver is your best option if it is available. ASIO does not guarantee low latency, but it allows for it if the audio card designers take advantage of its possibilities.

- Using a card via an ASIO driver can give latency figures as low as 3ms.
- When you use ASIO, only one program at a time can access the card.
- Note that to use ASIO you need to add an ASIO driver file to the ASIO Drivers folder in your Reason folder.
If several program take advantage of ASIO this means you will have to duplicate this driver to each program's ASIO Driver folder.

★ **More information about ASIO can be found on Steinberg Media Technologies' web pages, www.steinberg.net.**



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→ MIDI Implementation

About This Chapter

This chapter briefly describes the way various MIDI messages are implemented in Reason. It is mainly intended for those who control the rack directly via MIDI, but direct MIDI input can also be put to good use when recording into the sequencer.

The basics on how to send MIDI to Reason is described on [page 43](#) and [page 45](#). This chapter only deals with the details on various MIDI messages.

MIDI Direct Control vs. MIDI Remote

Please do not confuse direct MIDI control of devices with MIDI Remote.

- Direct MIDI Control uses a fixed set of MIDI messages for each device whereas MIDI Remote requires you to define which MIDI message to use for a certain control.
- MIDI Remote can not be recorded into the sequencer. If you send MIDI Controller messages via the sequencer input, they get recorded along with all other MIDI data, just as if you moved the controller with the mouse.

ReWire vs. Regular MIDI

You can use ReWire 2 to send MIDI messages to Reason. This uses exactly the same MIDI implementation as regular MIDI.

The MIDI Implementation Charts

In your program folder you will find a document called MIDI Implementation Charts.pdf. This contains tables of how all MIDI messages are implemented in various devices. Below follows a summary of various MIDI messages and their use.

How various MIDI messages are Implemented

Notes

The machines that receive MIDI notes are:

- Mixer 14:2 (for muting, soloing and activating EQ).
- Subtractor
- NN-19
- Redrum
- Dr. Rex
- NN-Xt
- Malström
- RV7000
- BV512
- ECF-42

The exact ranges and usage can be found in the MIDI Implementation Charts.

Controllers

Reason makes heavy use of MIDI controllers. Practically all controls on all devices can be controlled via MIDI.

The exact implementation of MIDI controllers for each device can be found in the MIDI Implementation Charts.

Modulation wheel, Expression and Breath can on some devices be routed to various controls by use of the device's front panel.

Pitch Bend

This is implemented on all devices where it makes sense to be able to bend notes via MIDI. Where pitch bend is implemented, there is a bend range control on the front panel.

Aftertouch

This is implemented on Subtractor, Malström, NN-19 and NN-Xt. It can be used to modulate various parameters.

Program Change and Polyphonic Aftertouch

These two MIDI messages are not implemented in any device.



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